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Total Assignments : 1

Serial Number : 09/157884 | Patent Number :
Filling Date : **09/21/1998** | Issue Date : **00/00/0000**
Related Patent Numbers:
Title : **MULTIMEDIA COMMUNICATIONS SOFTWARE WITH NETWORK STREAMING AND MULTI-FORMAT CONFERENCING**
Applicant(s) : VEGA-GARCIA, ANDRES || HAN, MU || RYAN, DON || PFENNING, THOMAS || BYRISETTY, RAJEEV || SOLOMON, STEFAN
Reel/Frame(s) : 009655/0763

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Reel : 009655 | Frame : 0763 | Mailroom Date : 12/11/1998 | New App Fil Rcpt Date : 00/00/0000
Assignments Date : 12/28/1998 | Date Mailed : 04/01/1999
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SOLOMON, STEFAN	11/19/1998		
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Brief :
ASSIGNMENT OF ASSIGNOR'S INTEREST (SEE DOCUMENT FOR DETAILS).

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Serial# Search **or Patent#** Search **or Reel/Frame#** Search

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Conference

709/204

379/93.21, 158, 202.1, 205.01

455/416

370/260, 261, 262, 263

375/147

audio G.711, G.723.1

video H.261, H.263

demultiplexer

receiver

audio-video conference

	Hits	Search Text
1	1306	345/156.CCLS. or 345/157.ccls. or 345/158.ccls. or 345/159.ccls.
2	801	379/93.21.ccls. or 379/202.ccls. or 379/96.ccls. or 379/88.13.ccls. or 379/93.09.ccls.
3	511	348/15.ccls. or 348/330.ccls. or 348/329.ccls. or 348/972.ccls. or 348/118.ccls.
4	818	455/6.3.ccls. or 455/340.ccls. or 455/825.ccls. or 455/24.ccls. or 455/825.25.ccls. or 455/31.1.ccls.
5	319	709/204.ccls. or 379/158.ccls.
6	3577	5 or 6 or 7 or 8 or 9
7	0	(conferenc$ and playback and multimedia and (realtime adj transport adj protocol)).detd.
8	177	(conferenc$ and playback and multimedia).detd.
9	46	12 and 10
10	0	(((receiv$ adj3 payload) same decoder same demultiplexer same mixer) and (multimedia or (audio and video)) and (type or format) and conferenc$).detd.
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12	0	(((payload) and decoder and demultiplexer and mixer) and (multimedia or (audio and video)) and (type or format) and conferenc$).detd.
13	6	(((stream or data) and decoder and demultiplexer and mixer) and (multimedia or (audio and video)) and (type or format) and conferenc$).detd.
14	8	((((DEMULTIPLEXER AND $CONFERENC$ AND MIXER AND DECODER) AND (MULTIMEDIA OR (AUDIO AND VIDEO))) AND (FORMAT OR TYPE)) AND (STREAM OR DATA)).DETD.

	Hits	Search Text
15	4	(("5684527") or ("5680512") or ("5636186") or ("4667088") or ("8301705")).PN.
16	17	(("6073192") or ("6052706") or ("6026097") or ("5991308") or ("5825680") or ("5805854") or ("5802281") or ("5708961") or ("5706048") or ("5687095") or ("5629736") or ("5623312") or ("5461679") or ("5267334") or ("5212742") or ("4679191") or ("4597077")).PN.
17	1	("5867494").PN.

	Hits	Search Text
1	28	(CONFERENC$ AND ((MULTIPLE? OR PLURALITY OR TWO) ADJ (PROTOCOL? OR FORMAT? OR STANDARD?))).bsum.
2	692	(((MULTIPLE? OR PLURALITY OR TWO) ADJ (PROTOCOL? OR FORMAT? OR STANDARD?))).bsum.
3	257	((MULTIPLE? OR PLURALITY OR TWO) ADJ (STANDARD?)).BSUM.
4	0	3 and 4
5	0	2 and 4
6	33	((h adj (261 or 263)) same (reassembl$ or align$ or realign$ or refram$ or combin$ or recombin$ or duplicat$ or (compensat$ or correct$))).detd.
7	261	((MULTIPLE? OR PLURALITY OR TWO) ADJ (STANDARD? or itu or ccit)).BSUM.
8	1	("5841763").PN.
9	2	((MULTIPLE? OR PLURALITY OR TWO) ADJ (ITU adj2 STANDARD?)).BSUM.
10	2	9 and (REFRAM$ OR REASSEMBL$ OR DUPLICAT$ OR RECOMBIN$ OR COMPENSAT$ OR CORRECT$ OR COMBIN$)
11	1	("5473366").PN.
12	1	(5473366.pn. AND (REFRAM$ OR REASSEMBL$ OR DUPLICAT$ OR RECOMBIN$ OR COMPENSAT$ OR CORRECT$ OR COMBIN$))
13	1	(("5473366".PN.) AND (REFRAM$ OR REASSEMBL$ OR DUPLICAT$ OR RECOMBIN$ OR COMPENSAT$ OR CORRECT$ OR COMBIN$ or split$ or realign$ or duplicat$))
14	4	(("6058465") or ("5978838") or ("5860158") or ("5838984")).PN.

	Hits	Search Text
15	12	(("6034968") or ("5999525") or ("5910827") or ("5867494") or ("5859904") or ("5751700") or ("5673080") or ("5625407") or ("5555017") or ("5541852") or ("5473366") or ("5453780")).PN.
16	12	(("6034968") or ("5999525") or ("5910827") or ("5867494") or ("5859904") or ("5751700") or ("5673080") or ("5625407") or ("5555017") or ("5541852") or ("5473366") or ("5453780") or ("5751700") or ("5473366") or ("5453780")).PN.
17	1	("6075554").PN.
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19	1	("6075554").PN.
20	27	(("6075554") or ("6034968") or ("5982459") or ("5982418") or ("5953050") or ("5936945") or ("5926208") or ("5859904") or ("5838664") or ("5835129") or ("5821987") or ("5790178") or ("5784457") or ("5777663") or ("5764278") or ("5734415") or ("5706290") or ("5689300") or ("5612745") or ("5606554") or ("5600646") or ("5541640") or ("5491515") or ("5477546") or ("5453780") or ("5363378") or ("5235420")).PN.
21	1	("5838664").PN.
22	0	(vega-garcia).in.
23	2206	(han, mu).in.
24	1867	(HAN).IN.
25	0	conferenc.bsum. and 24

	Hits	Search Text
26	21	conferenc$.bsum. and 24
27	0	(han adj mu).in.
28	0	(ryan adj don).in.
29	21	(pfenning).in.
30	107	(rajeev).inv.
31	2607	(RAJEEV oe solomon or pfenning or han or vega-garcia).INV.
32	1	("5541852").PN.
33	1	(h adj (263 or 261) and g adj (723$ or 711)).detd.
34	1	((G ADJ ("723"$ OR "711")) AND (H ADJ ("263" OR "261")))
35	1	((G ADJ ("711$" OR "723"$)) AND (H ADJ ("261$" OR "263$")))
36	0	((multiple or plurality or two) adj2 (itu or ccit) adj2 standard).bsum.
37	2	((multiple or plurality or two) adj2 (itu or ccit) adj2 standard$).bsum.
38	1	(((MULTIPLE OR PLURALITY OR TWO) ADJ2 (CCIT OR ITU)) ADJ2 STANDARD$).detd.

	Search Text
1	(709/204.ccls.)
2	(conferenc$ and real-time).bsum.
3	(REAL-TIME AND CONFERENC$ and (render$ or playback)).BSUM.
4	((REAL adj TIME AND CONFERENC$) AND (demux or demultiplexer)).BSUM.
5	(((DEMUX OR DEMULTIPLEXER) AND CONFERENC$) AND (REAL-TIME)).BSUM.
6	((REAL-TIME or rtp) AND $CONFERENC$).detd.
7	345/$.ccls. and 5
8	((render$ or playback AND CONFERENC$) AND (REAL ADJ TIME)).BSUM.
9	((REAL ADJ TIME) AND CONFERENC$ AND ((PLAYBACK OR RENDER$) same (audio or video))).BSUM.
10	((CONFERENC$ AND (REAL ADJ TIME)) AND ((PLAYBACK OR RENDER$) SAME (AUDIO OR VIDEO)) same (demultiplexer or demux)).detd.
11	((CONFERENC$ AND (REAL-TIME)) AND ((PLAYBACK OR RENDER$) SAME (AUDIO OR VIDEO)) same (demultiplexer or demux)).detd.
12	((CONFERENC$ AND (REAL-TIME)) AND ((PLAYBACK OR RENDER$) SAME (AUDIO OR VIDEO)) same (demultiplexer or demux))
13	((REAL-TIME or rtp) AND CONFERENC$).bsum.
14	((RTP OR REAL-TIME) AND CONFERENC$ and (payload adj3 handler?) and decoder? and mixer and (demux or demultiplexer) and (render$ or output$ or play$))
15	((RTP OR REAL-TIME) AND CONFERENC$ and decoder? and mixer and (demux or demultiplexer) and (render$ or output$ or play$))
16	((RTP OR REAL-TIME) AND CONFERENC$ and (RENDER$ OR PLAY$ OR OUTPUT$)).bsum.
17	16 and (DEMUX? OR DEMULTIPLEXER?)
18	((DEMUX$ OR DEMULTIPLEX$ or multiplex$) AND 16)
19	(directshow same microsoft$).detd.

US006122665A

United States Patent [19]

Bar et al.

[11] **Patent Number: 6,122,665**

[45] **Date of Patent: Sep. 19, 2000**

[54] **COMMUNICATION MANAGEMENT SYSTEM FOR COMPUTER NETWORK-BASED TELEPHONES**

[75] Inventors: **Eitan Bar**, Tzoran; **Mordechai Nisani**, Tel Aviv, both of Israel

[73] Assignee: **STS Software System Ltd.**, Tel Aviv, Israel

[21] Appl. No.: **09/140,453**

[22] Filed: **Aug. 26, 1998**

[51] **Int. Cl.**[7] **G06F 15/173**

[52] **U.S. Cl.** **709/224**; 709/204; 379/88

[58] **Field of Search** 709/204, 205, 709/206, 207, 224, 227; 345/302; 379/67, 70, 88

[56] **References Cited**

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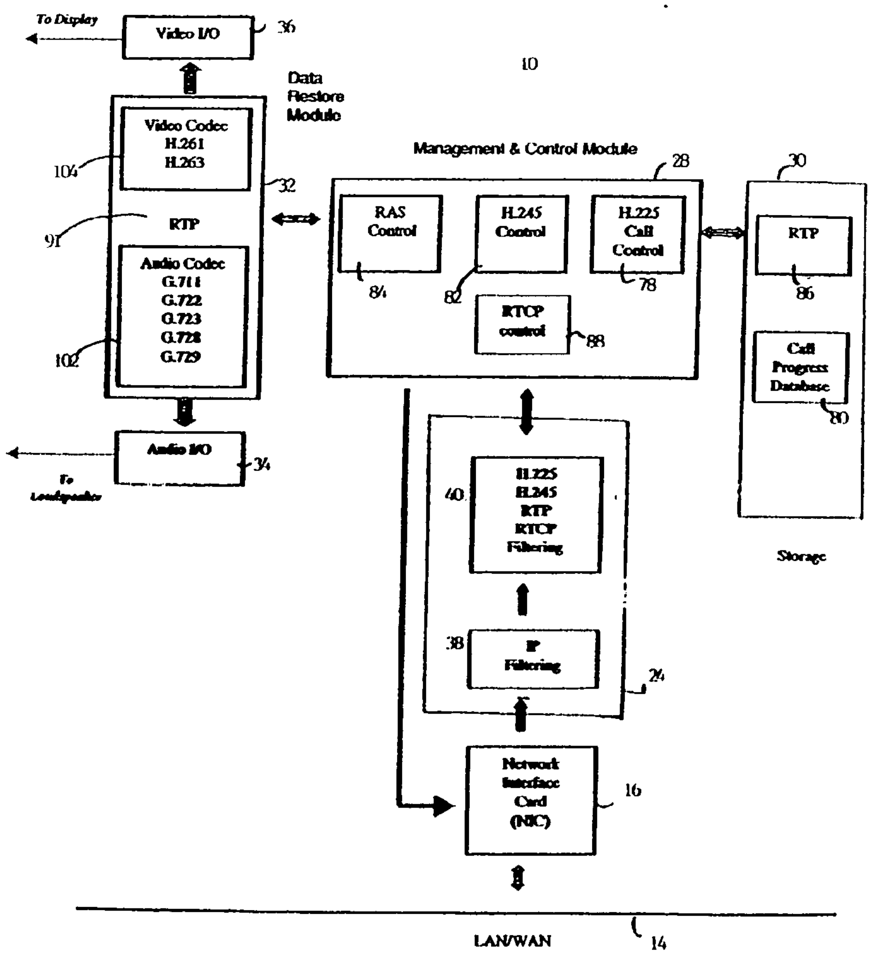
5,351,243 9/1994 Kalkunte et al. 370/475
5,430,709 7/1995 Galloway 370/241
5,689,641 11/1997 Ludwig et al. 395/200.71
5,717,879 2/1998 Moran et al. 345/330
5,742,833 4/1998 Dea et al. 340/825.02
5,848,233 12/1998 Radia et al. 713/201
5,964,839 10/1999 Johnson et al. 702/187

Primary Examiner—Dung C. Dinh
Attorney, Agent, or Firm—Mark M. Friedman

[57] **ABSTRACT**

A system and a method for monitoring a computer network to detect data packets including audio or video data, such packets being part of a communication session, for storing these packets and for reconstructing the communication session upon request.

23 Claims, 8 Drawing Sheets



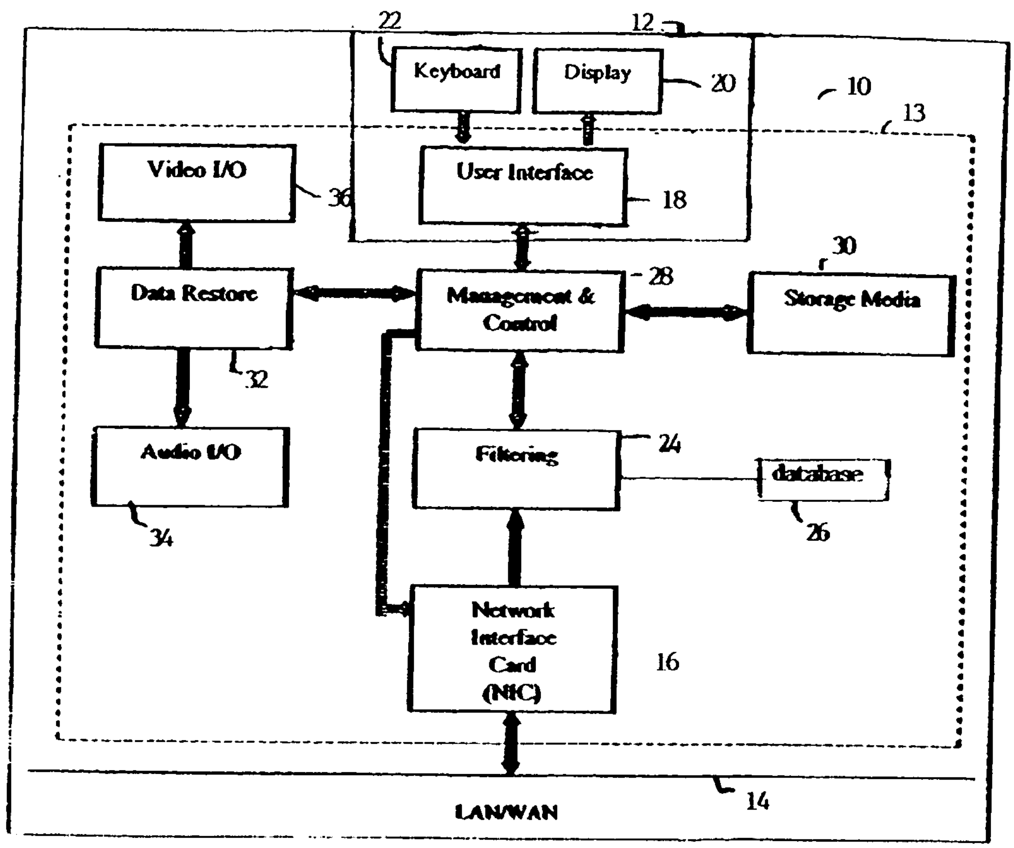

22
12
Keyboard
Display
20
10
13
Video I/O
36
User Interface
18
30
Data Restore
Management & Control
28
Storage Media
32
Audio I/O
Filtering
24
database
34
26
Network Interface Card (NIC)
16
LAN/WAN
14

FIG. 1

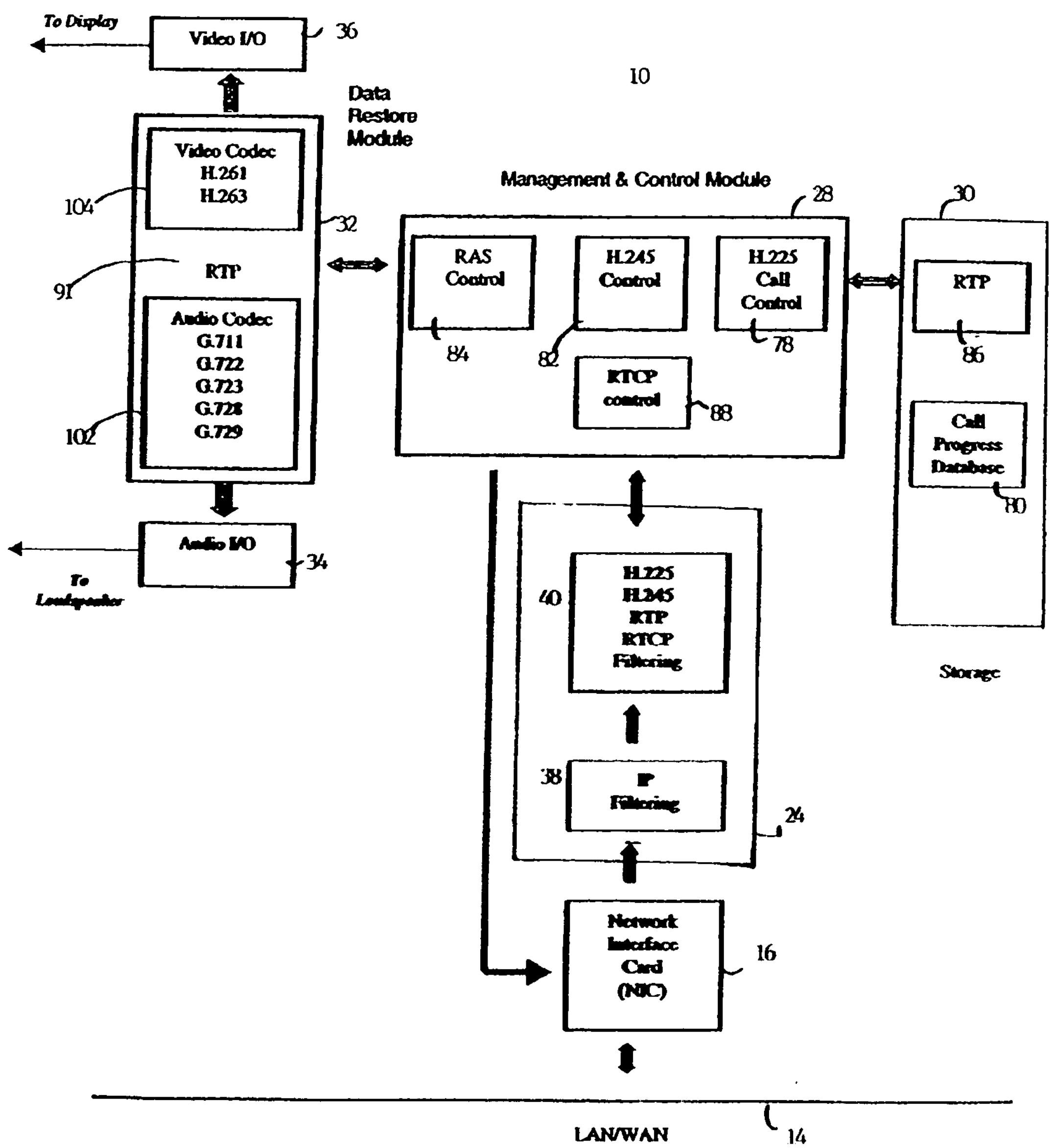

To Display
Video I/O
36
10
Data Restore Module
Video Codec H.261 H.263
104
32
RTP
91
Audio Codec G.711 G.722 G.723 G.728 G.729
102
Management & Control Module
28
30
RAS Control
84
H.245 Control
82
H.225 Call Control
78
RTCP control
88
RTP
86
Call Progress Database
80
Audio I/O
34
To Loudspeaker
40
H.225 H.245 RTP RTCP Filtering
Storage
38
IP Filtering
24
Network Interface Card (NIC)
16
LAN/WAN
14

FIG. 2

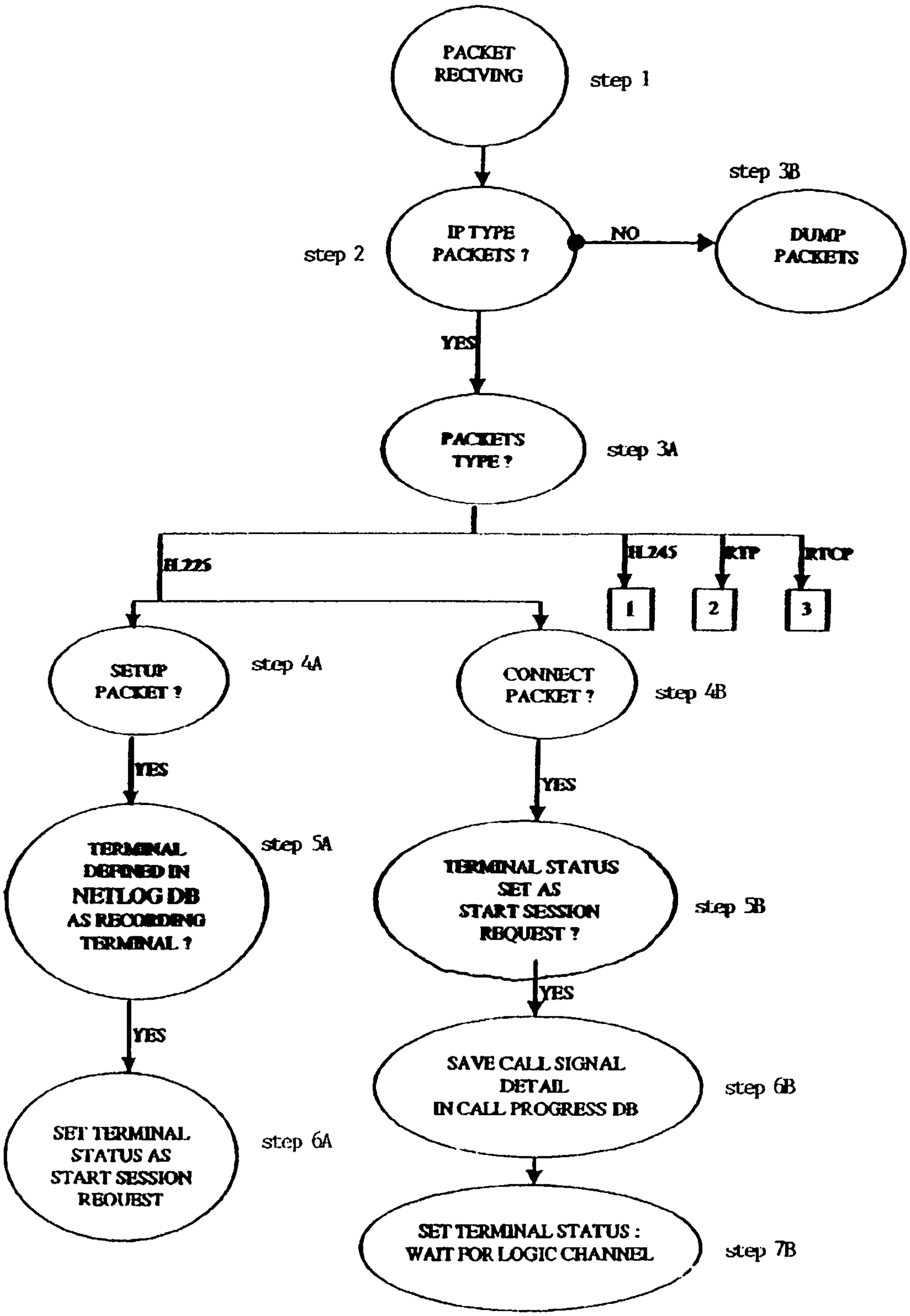

PACKET RECIVING
step 1
IP TYPE PACKETS ?
step 2
NO
step 3B
DUMP PACKETS
YES
PACKETS TYPE ?
step 3A
H.225
H.245
RTP
RTCP
1
2
3
SETUP PACKET ?
step 4A
CONNECT PACKET ?
step 4B
YES
YES
TERMINAL DEFINED IN NETLOG DB AS RECORDING TERMINAL ?
step 5A
TERMINAL STATUS SET AS START SESSION REQUEST ?
step 5B
YES
YES
SET TERMINAL STATUS AS START SESSION REQUEST
step 6A
SAVE CALL SIGNAL DETAIL IN CALL PROGRESS DB
step 6B
SET TERMINAL STATUS : WAIT FOR LOGIC CHANNEL
step 7B

FIG. 3A

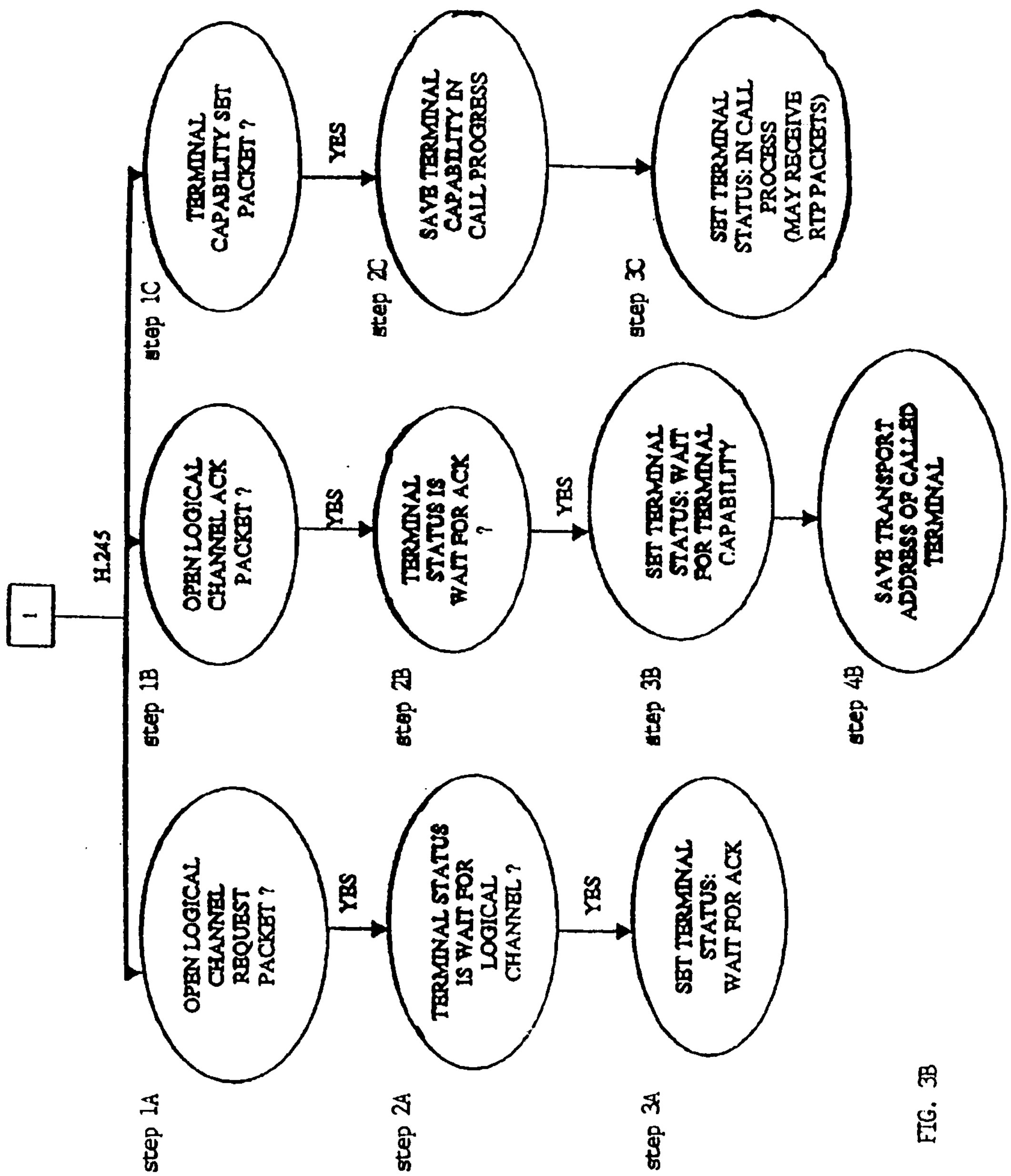

1
H.245
step 1A
OPEN LOGICAL CHANNEL REQUEST PACKET ?
YES
step 2A
TERMINAL STATUS IS WAIT FOR LOGICAL CHANNEL ?
YES
step 3A
SET TERMINAL STATUS: WAIT FOR ACK
step 1B
OPEN LOGICAL CHANNEL ACK PACKET ?
YES
step 2B
TERMINAL STATUS IS WAIT FOR ACK ?
YES
step 3B
SET TERMINAL STATUS: WAIT FOR TERMINAL CAPABILITY
step 4B
SAVE TRANSPORT ADDRESS OF CALLED TERMINAL
step 1C
TERMINAL CAPABILITY SET PACKET ?
YES
step 2C
SAVE TERMINAL CAPABILITY IN CALL PROGRESS
step 3C
SET TERMINAL STATUS: IN CALL PROCESS (MAY RECEIVE RTP PACKETS)

FIG. 3B

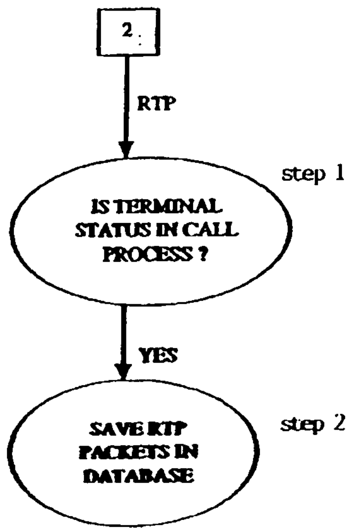


FIG. 3C

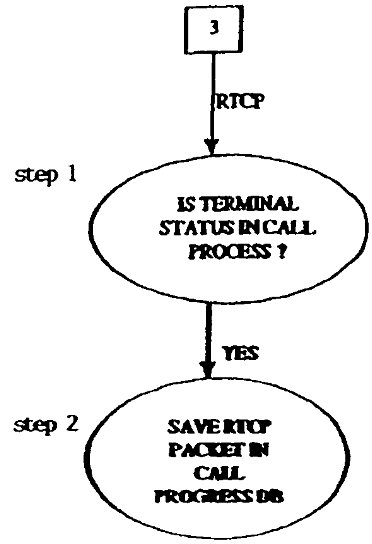


FIG. 3D

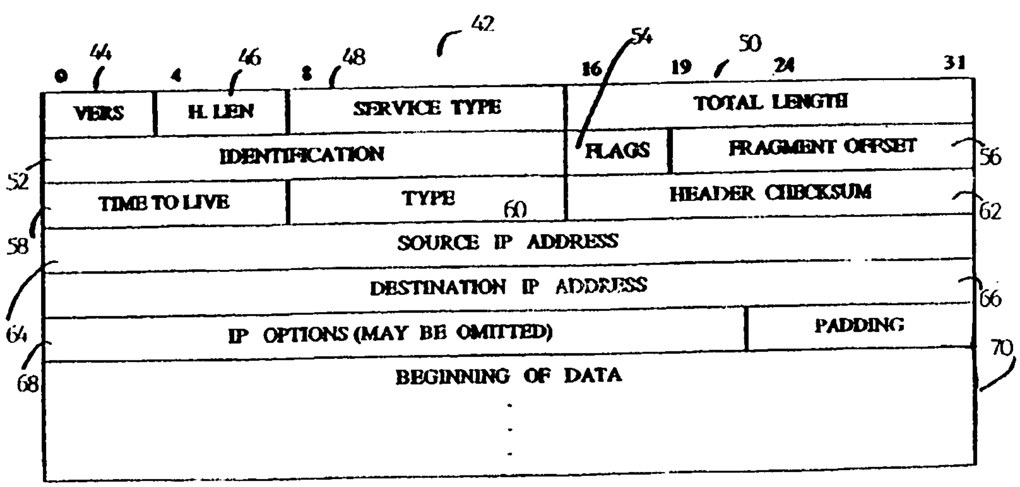


FIG. 4A

H.225 and H.245 packet

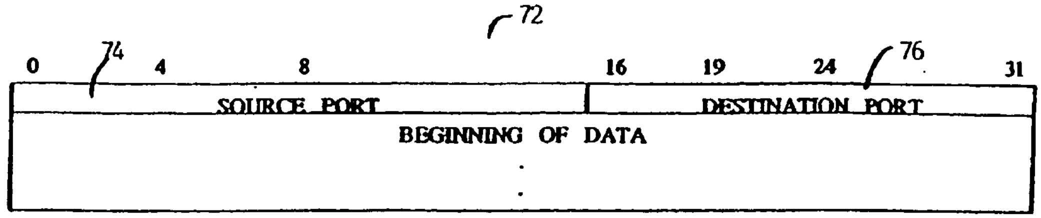


FIG. 4B

RTP packet

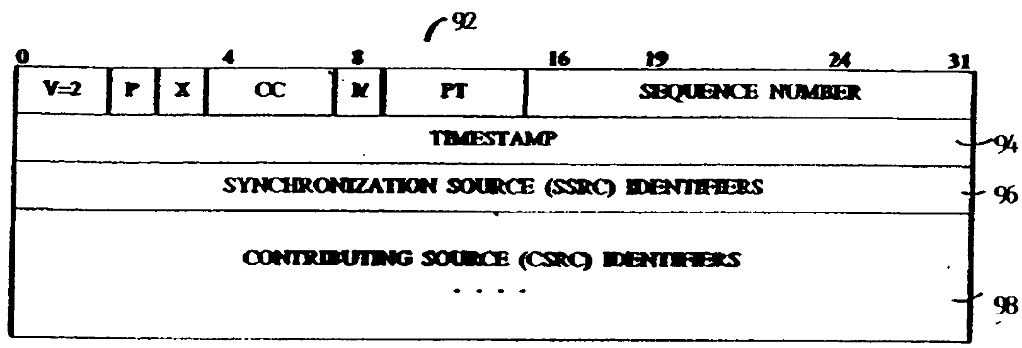


FIG. 4C

RTCP packet

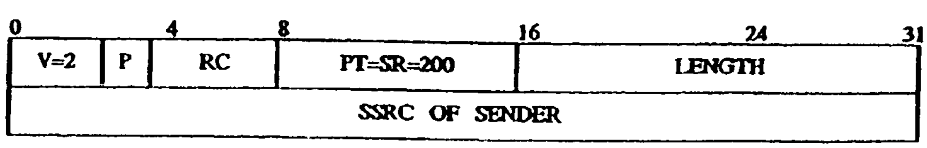


FIG. 4D

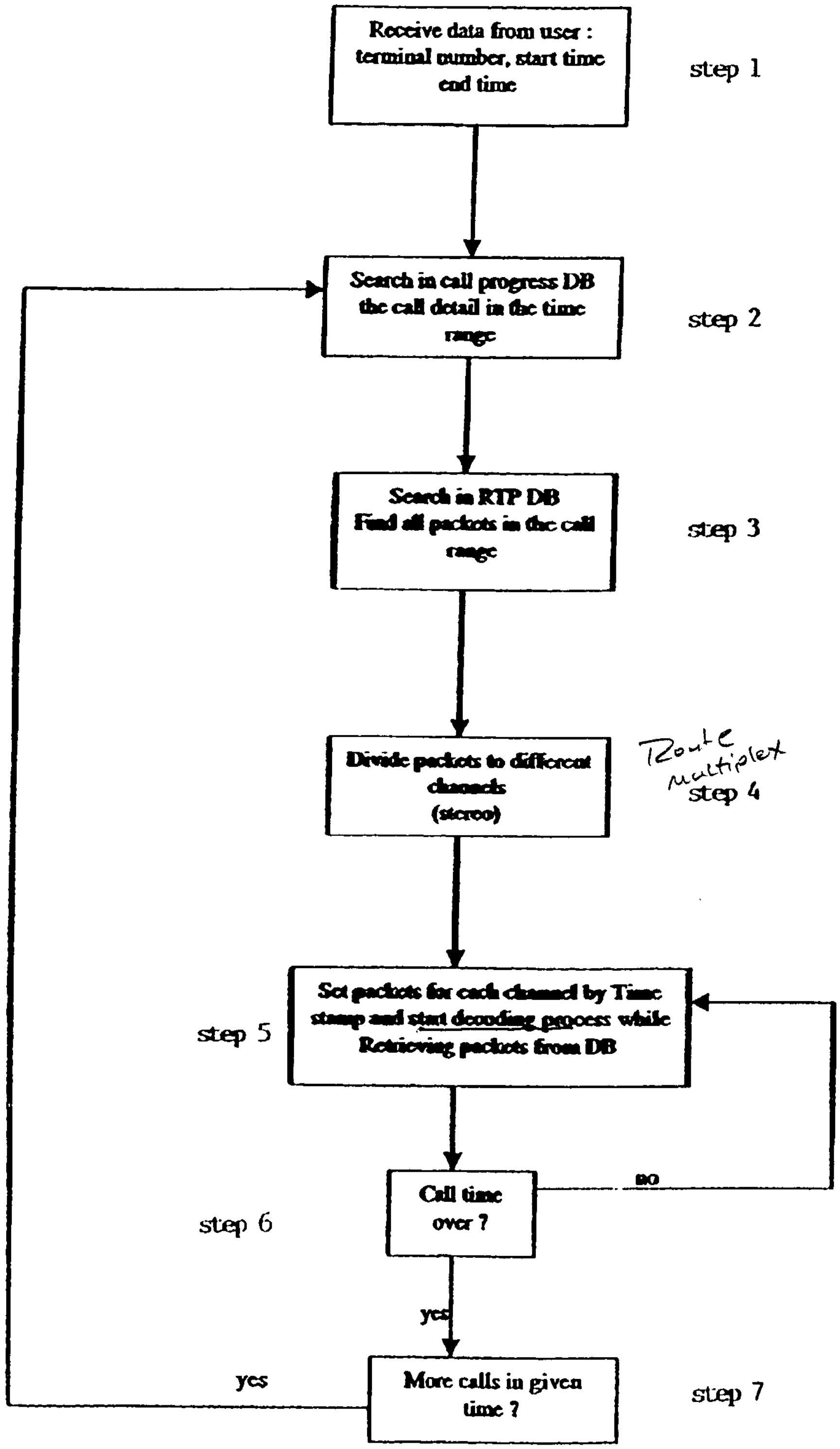
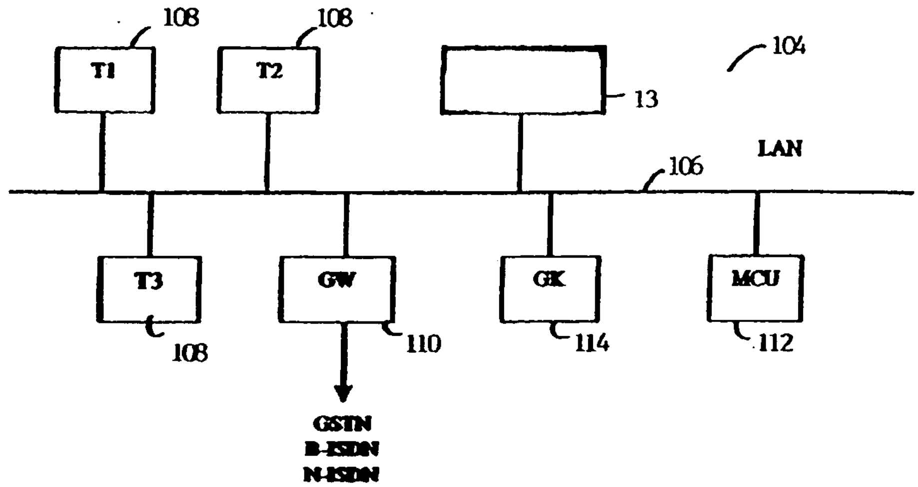

Receive data from user : terminal number, start time end time
step 1
Search in call progress DB the call detail in the time range
step 2
Search in RTP DB Find all packets in the call range
step 3
Divide packets to different channels (stereo)
Route multiplex
step 4
Set packets for each channel by Time stamp and start decoding process while Retrieving packets from DB
step 5
Call time over ?
step 6
no
yes
More calls in given time ?
step 7
yes

FIG. 5


108
T1
108
T2
13
104
LAN
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T3
108
GW
110
GK
114
MCU
112
GSTN
B-ISDN
N-ISDN

FIG. 6

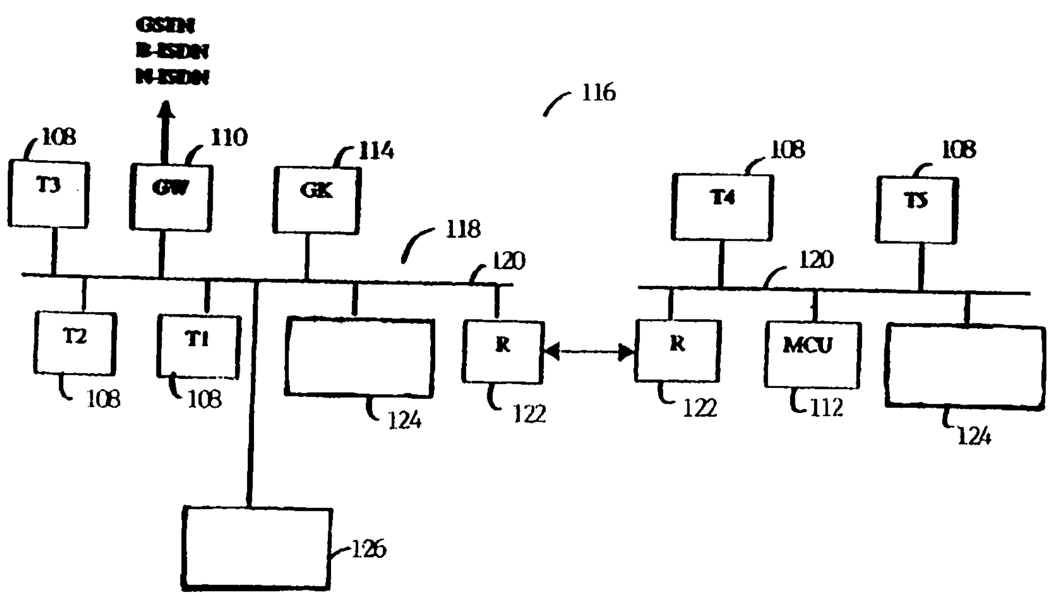
GSTN
B-ISDN
N-ISDN
116
108
T3
110
GW
114
GK
118
120
108
T4
108
T5
120
T2
108
T1
108
124
R
122
R
122
MCU
112
124
126

FIG. 7

COMMUNICATION MANAGEMENT SYSTEM FOR COMPUTER NETWORK-BASED TELEPHONES

FIELD AND BACKGROUND

6-12

The present invention is of a method and a system for the management of communication sessions for computer network-based telephone communication, and in particular for the identification of packets containing audio and/or video data, for the storage of these packets, and for the reconstruction of selected communication sessions for audio and/or video display as needed.

The integration of the computer into office communication systems has enabled many functions previously performed by separate devices to be combined into a single management system operated through a computer. For example, computer-based voice logging systems enable a computer to receive voice communication through a hardware connection to the regular telephony network, to record either a conversation, in which at least two parties converse, or a message from at least one party to one or more parties, and to replay these recorded conversations or messages upon request. These voice logging systems can replace mechanical telephone answering machines.

The computer logging systems have many advantages over the mechanical answering machines. For example, the voice messages can be stored in a computer-based storage medium, such as a DAT cassette, which has a greater storage capacity than regular audio cassettes. Furthermore, the stored voice messages can be organized in a database, such that the messages can retrieved according to time, date, channel, dialed number or caller identification, for example. Such organization is not possible with a mechanical telephone answering machine. Thus, computer logging systems for voice messages have many advantages over mechanical answering machines.

Unfortunately, currently available computer logging systems have the disadvantage of being unable to record telephone communication sessions, whether conversations or messages, for voice communication being performed through a LAN (local area network) or a WAN (wide area network). Although these logging systems can play back voice messages to a remote user through a LAN, for example, they cannot record such a message if it is transmitted by a LAN-based telephone. Such LAN and WAN based telephone communication has become more popular recently, since it enables telephone communication to be performed between various parties at physically separated sites without paying for local regular telephony network services, thereby saving money.

Furthermore, LAN and WAN based telephone communication also facilitates the transmission of video as well as audio information. Video information certainly cannot be recorded by currently available computer logging systems. Thus, the inability of computer logging systems to record telephone communication sessions for telephone communication being performed through a LAN or a WAN, including both video and audio data, is a significant disadvantage of these systems.

There is therefore a need for, and it would be highly advantageous to have, a system and a method for recording telephone communication sessions performed over a computer network such as a LAN or a WAN, which would record both audio and video information, organize such information, and then display such information upon request.

SUMMARY OF THE INVENTION

It is one object of the present invention to provide a system and a method for recording communication sessions performed over a computer network.

It is another object of the present invention to provide such a system and method for analyzing data transmitted over the computer network in order to detect audio and video data for recording.

It is still another object of the present invention to provide such a system and method for displaying recorded video and audio data upon request.

It is yet another object of the present invention to provide such a system and method for analyzing, recording and displaying communication sessions conducted with a LAN-based telephone system.

These and other objects of the present invention are explained in further detail with regard to the drawings, description and claims provided below.

The present invention provides a system and a method for analyzing data packets on a computer network, for selectively recording audio and video data packets, for organizing this stored information and for displaying the stored information upon request, such that communication sessions with computer network-based "telephone" systems can be logged.

According to the teachings of the present invention, there is provided a system for managing a communication session over a computer network, the system comprising: (a) a network connector for connecting to the computer network and for receiving data packets from the computer network; (b) a filtering unit for filtering the data packets and for accepting the data packets substantially only if the data packets contain data selected from the group consisting of audio data and video data, such that the data packets form at least a portion of the conununication session and such that the data packets are selected data packets; (c) a management unit for receiving the selected data packets and for storing the selected data packets, such that the selected data packets are stored data packets; and (d) a storage medium for receiving and for storing the stored data packets from the management unit, such that the at least a portion of the communication session is stored. 32-37

Preferably, the system further comprises (e) a data restore unit for retrieving and displaying the at least a portion of the communication session, the data restore unit requesting the data packets from the storage medium through the management unit, and the data restore unit reconstructing the data packets for displaying the at least a portion of the communication session. 2/44-50

More preferably, the data restore unit further comprises a communication session display unit for displaying the at least a portion of the communication session. Most preferably, the communication session display unit is selected from the group consisting of a video unit and an audio unit.

According to preferred embodiments of the present invention, the system further comprises (f) a database connected to the filtering unit for storing filtering information, the filtering information including at least one IP address of a party whose communication sessions are monitored; wherein the filtering unit accepts the data packets according to the filtering information, such that the filtering unit substantially only accepts the data packets if the data packets fulfill the filtering information. 58-61

Preferably, the system further comprises (g) a user computer for receiving at least one command of a user and for

displaying information to the user, such that the user determines the filtering information according to the at least one command of the user.

More preferably, the computer network is selected from the group consisting of a LAN (local area network) and a WAN (wide area network). Most preferably, the computer network is a LAN (local area network).

According to further preferred embodiments of the present invention, the LAN is divided into at least two segments, the system further comprising: (h) a local management unit for each segment, the local management unit including the filtering unit and the management unit; and (i) a central management unit for controlling the local management units, the central management unit controlling storage in the storage medium.

Preferably, the network connector is a network interface card.

According to another embodiment of the present invention, there is provided a method for storing at least a portion of a communication session performed on a computer network, the communication session being performed between a packet source and a packet destination, the steps of the method being performed by a data processor, the method comprising the steps of: (a) receiving a data packet from the packet source on the computer network; (b) analyzing the data packet to determine if the data packet is an IP packet; (c) if the data packet is the IP packet, filtering the IP packet to determine a type of the IP packet; and (d) storing the IP packet to form a stored data packet according to the type, such that the stored data packet forms at least a portion of the communication session. Preferably, the step of analyzing the data packet is performed by examining a header of the data packet.

3/25-34

According to a preferred embodiment of the present invention, the step of filtering the IP packet is performed by examining the header of the IP packet.

Preferably, the step of filtering the IP packet further comprises the steps of: (i) examining the header of the IP packet to determine an IP address of the packet source; (ii) determining if the IP address is a recorded IP address; (iii) passing the IP packet to form a passed IP packet substantially only if the IP address is the recorded IP address; and (iv) alternatively, dumping the IP packet.

More preferably, the step of determining if the IP address is the recorded IP address is performed by comparing the IP address to a list of IP addresses from packet sources, such that if the IP address is included in the list, the IP address is the recorded IP address.

Also preferably, the step of filtering the IP packet further comprises the steps of: (v) determining whether the passed IP packet is an H.225 packet, a H.245 packet, an RTP packet or an RTCP packet; (vi) if the type of the passed IP packet is the H.225 packet, determining whether the H.225 packet is a setup packet or a connect packet; (vii) if the H.225 packet is the setup packet, setting a status flag as "start session request"; (viii) alternatively, if the H.225 packet is the connect packet and the status flag is "start session request", storing at least one detail of the communication session; and (ix) setting the status flag as "wait for logic channel".

50-61

More preferably, the step of filtering the IP packet further comprises the steps of: (x) alternatively, if the type of the passed IP packet is the H.245 packet, determining whether the H.245 packet is an open logical channel request packet, an open logical channel acknowledgment packet or a terminal capability set packet; (xi) if the H.245 packet is the open logical channel request packet and the status flag is "wait for logic channel", setting the status flag as "wait for acknowledgment"; (xii) alternatively, if the H.245 packet is the open logical channel acknowledgment packet and the status flag is "wait for acknowledgment", performing the steps of: (A) setting the status flag as "wait for terminal capability"; and (B) saving a transport address of the destination of the communication session; and (xiii) also alternatively, if the H.245 packet is the terminal capability set packet, performing the steps of: (A) storing a capability of the packet destination from the terminal capability packet; and (B) setting the status flag as "in call process".

Most preferably, if the status flag is "in call process" and the type of the passed IP packet is the RTP packet, the RTP packet is stored. Also most preferably, if the status flag is "in call process" and the type of the passed IP packet is the RTCP packet, the RTCP packet is stored. * 14-18

According to another preferred embodiment of the present invention, the method further comprises the steps of: (e) retrieving the stored data packet to form a retrieved data packet; and (f) reconstructing at least a portion of the communication session according to the retrieved data packet.

3/50-61
4/19-24, 31-36

Preferably, the step of retrieving the data packet includes the steps of: (i) receiving a source IP address of the packet source, a start time of the communication session, and an end time of the communication session; and (ii) selecting at least one communication session according to the source IP address, the start time and the end time.

Also preferably, the step of reconstructing at least a portion of the communication session includes displaying audio data.

Alternatively and also preferably, the step of reconstructing at least a portion of the communication session includes displaying video data.

More preferably, the step of reconstructing at least a portion of the communication session further comprises the steps of: (i) retrieving substantially only RTP packets; (ii) examining a header of the RTP packets to determine a time stamp for each of the RTP packets; and (iii) displaying the RTP packets in an order according to the time stamp.

Hereinafter, the term "communication session" includes both a conversation, in which at least two parties converse by exchanging audio and/or video information in "real time", and a message, in which at least one party records such audio and/or video information for reception by at least one other party at a later date.

37-48 *

Hereinafter, the term "Internet" is used to generally designate the global, linked web of thousands of networks which is used to connect computers all over the world. As used herein, the term "intranet" includes other types of computer networks, such as LAN (local area networks) or WAN (wide area networks). The term "computer network" includes any connection between at least two computers which permits the transmission of data, including both Internet and intranet. The term "regular telephony network" includes POTS (plain old telephone system) and substantially any other type of telephone network which provides services through a regular telephone services provider, but which specifically excludes audio and/or video communication performed through any type of computer network.

54-57

Hereinafter, the term "computer" includes, but is not limited to, personal computers (PC) having an operating system such as DOS, Windows™, OS/2™ or Linux; Mackintosh™ computers; computers having JAVA™-OS as the operating system; and graphical workstations such as the

computers of Sun Microsystems™ and Silicon Graphics™, and other computers having some version of the UNIX operating system such as AIX or SOLARIS™ of Sun Microsystems™; or any other known and available operating system. Hereinafter, the term "Windows™" includes but is not limited to Windows95™, Windows 3.x™ in which "x" is an integer such as "1", Windows NT™, Windows98™, Windows CE™ and any upgraded versions of these operating systems by Microsoft Inc. (Seattle, Wash., USA).

4/63-5/9

Hereinafter, the term "logging" refers to the process of analyzing data packets on a network to locate audio and/or video data, and of recording such data in an organized system. Hereinafter, the term "display" includes both the visual display of video data, and the production of sound for audio data.

render 13-15

BRIEF DESCRIPTION OF THE DRAWINGS

The invention is herein described, by way of example only, with reference to the accompanying drawings, wherein:

FIG. 1 is a schematic block diagram of an exemplary communication session monitoring system according to the present invention;

FIG. 2 is a schematic block diagram of the software modules required for operating the system of FIG. 1;

FIGS. 3A–3D are flowcharts of exemplary filtering and recording methods according to the present invention;

FIGS. 4A–4D are schematic block diagrams showing the headers of H.225 (FIG. 4A), H.245 (FIG. 4B), RTP (FIG. 4C) and RTCP (FIG. 4D) packets, as they relate to the present invention;

FIG. 5 is a flowchart of an exemplary communication session playback method according to the present invention;

FIG. 6 is a schematic block diagram of an exemplary first embodiment of a basic system using the communication session monitoring system of FIGS. 1 and 2 according to the present invention; and

FIG. 7 is a schematic block diagram of an exemplary second embodiment of a zone system according to the present invention.

DESCRIPTION OF BACKGROUND ART

The following description is intended to provide a description of certain background methods and technologies which are optionally used in the method and system of the present invention. The present invention is specifically not drawn to these methods and technologies alone. Rather, they are used as tools to accomplish the goal of the present invention, which is a system and a method for analyzing data packets on a computer network, for selectively recording audio and video data packets, for organizing this stored information and for displaying the stored information upon request, such that communication sessions with computer network-based "telephone" systems can be logged.

MO

The system and method of the present invention is particularly intended for operation with computer networks constructed according to the ITU-T Recommendation H.323 for visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service. Recommendation H.323 is herein incorporated by reference in order to further describe the hardware requirements and operating protocols for such computer networks, and is hereinafter referred to as "H.323".

H.323 describes terminals, equipment and services for multimedia communication over Local Area Networks (LAN) which do not provide a guaranteed quality of service. Computer terminals and equipment which fulfill H.323 may carry real-time voice, data and video, or any combination, including videotelephony.

The LAN over which such terminals communicate can be a single segment or ring, or optionally can include multiple segments with complex topologies. These terminals are optionally integrated into computers or alternatively are implemented in stand-alone devices such as videotelephones. Support for voice data is required, while support for general data and video data are optional, but if supported, the ability to use a specified common mode of operation is required, so that all terminals supporting that particular media type can communicate. The H.323 Recommendation allows more than one channel of each type to be in use. Other Recommendations in the H.323-Series which are also incorporated by reference include H.225.0 packet and synchronization; H.245 control, H.261 and H.263 video codecs; G.711, G.722, G.728, G.729, and G.723 audio codecs; and the T.120-Series of multimedia communications protocols.

5-20

channel/protocol

ITU-T Recommendation H.245.0 covers the definition of Media stream packetization and synchronization for visual telephone systems. ITU-T Recommendation H.245.0 defines the Control protocol for multimedia communications, and is hereinafter referred to as "H.245". H.245 is incorporated by reference as is fully set forth herein.

The logical channel signaling procedures of H.245 describes the content of each logical channel when the channel is opened. Procedures are provided for the communication of the functional capabilities of receivers and transmitters, so that transmissions are limited to information which can be decoded by the receivers, and so that receivers may request a particular desired mode from transmitters.

21-40

H.245 signaling is established between two endpoints: an endpoint and a multi-point controller, or an endpoint and a Gatekeeper. The endpoint establishes exactly one H.245 Control Channel for each call that the endpoint is participating in. The channel must then operate according to H.245. Support for multiple calls and hence for multiple H.245 Control Channels is possible.

multiple-point

The RAS signaling function uses H.225.0 messages to perform registration, admissions, bandwidth changes, status, and disengage procedures between endpoints and Gatekeepers. In LAN environments that do not have a Gatekeeper, the RAS Signaling Channel is not used. In LAN environments which contain a Gatekeeper, such that the LAN includes at least one Zone, the RAS Signaling Channel is opened between the endpoint and the Gatekeeper. The RAS Signaling Channel is opened prior to the establishment of any other channels between H.323 endpoints.

The call signaling function uses H.225.0 call signaling to establish a connection between two H.323 endpoints. The Call Signaling Channel is independent from the RAS Channel and the H.245 Control Channel. The Call Signaling Channel is opened prior to the establishment of the H.245 Channel and any other logical channels between H.323 endpoints. In systems that do not have a Gatekeeper, the Call Signaling Channel is opened between the two endpoints involved in the call. In systems which contain a Gatekeeper, the Call Signaling Channel is opened between the end point and the Gatekeeper, or between the endpoints themselves as chosen by the Gatekeeper.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

The present invention provides a system and a method for analyzing data packets on a computer network, for selec-

tively recording audio and video data packets, for organizing this stored information and for displaying the stored information upon request, such that communication sessions with computer network-based "telephone" systems can be logged.

The principles and operation of a method and a system according to the present invention may be better understood with reference to the drawings and the accompanying description.

Referring now to the drawings, FIG. 1 is a block diagram of an exemplary system for logging and displaying audio and/or visual data from communication sessions performed over a computer network. A computer logging system 10 features a user computer 12 connected to a communication session management unit 13. Communication session management unit 13 is in turn connected to an intranet 14 through a network interface card (NIC) 16.

User computer 12 includes a user interface 18, which is preferably a GUI (graphical user interface), which is displayed on a display unit 20. User interface 18 preferably enables the user to enter such information as the definition of the parties whose calls should to be monitored and/or logged, and which also preferably enables the user to enter at least one command for retrieving and displaying a communication session.

MO 21-25

Display unit 20 is preferably a computer monitor. The user is able to interact with user computer 12 by entering data and commands through a data entry device 22. Data entry device 22 preferably includes at least a keyboard or a pointing device such as a mouse, and more preferably includes both a keyboard and a pointing device. According to one preferred embodiment of the present invention, user computer 12 is a PC (personal computer). Alternatively and preferably, user computer 12 is a "thin client" such a net computer which is a computer able to communicate on an IP-based network. One example of such a net computer is the JavaStation™ (Sun Microsystems). The advantage of such net computers is that they allow the user to interact with complex, sophisticated software programs, yet generally do not have all of the powerful computing capabilities of currently available PC computers.

Intranet 14 could be a LAN or a WAN, for example. The connection between communication session management unit 13 and intranet 14 occurs through NIC 16. NIC 16 is preferably any standard, off-the-shelf commercial product which enables communication session management unit 13 to be connected to any suitable computer network (for example, Etherlink II ISA/PCMCIA Adapter or Etherlink III PCI Bus-Master Adapter (3c590) of 3-Com™, or NE2000 Adapter of Novell™ or any other such suitable product). Examples of such suitable computer networks include, but are not limited to, any standard LAN such as Ethernet (IEEE Standard 802.3), Fast Ethernet (IEEE Standard 802.10), Token Ring (IEEE Standard 802.5) and FDDI.

All data packet traffic on intranet 14 is passed to a filtering module 24 through NIC 16. As shown in more detail in FIG. 3 below, filtering module 24 screens the data packets in order to determine which data packets fulfill the following criteria. Briefly, the data packets should be IP packets with headers according to the H.225 and H.245 standards, indicating voice and/or video traffic. As noted previously, these standards define media stream packet construction and synchronization for visual telephone systems and the control protocol for multimedia communications.

datatypes MO

Filtering module 24 then preferably passes substantially only those data packets which meet these criteria to a management module 28. In the Zone Configuration of the system of the present invention, shown in FIG. 7 below, filtering module 24 preferably also transfers messages from other communication session management units.

7/56-8/4

Management module 28 receives the data packets passed through by filtering module 24, and analyzes the received data packets. Optionally and preferably, a database 26 stores such information as the IP addresses of parties whose communication sessions should be logged, as well as the conversion table associating each party with at least one IP address, for example. The stored list of IP addresses representing those parties whose calls should be logged is preferably user-defined. As used herein, the term "party" refers to a person or persons communicating through a computer network-based telephone system. The latter preferred requirement significantly reduces the amount of data stored by including only data which is of interest to the user. Management module 28 analyzes and manages data in accordance with the applicable H.225 and H.245 specifications, including the H.245 control function, RAS signaling function and call signaling function, substantially as described above in the "Description of the Background Art" section.

Management module 28 analyzes the packets in order to determine the specific communication session to which the data packets belong, the type of data compression being used (if any), and whether the data packets were sent from an IP address which should be monitored. Management module 28 must perform this analysis since filtering module 24 simply passes all data packets which fulfill the criteria described briefly above (see FIGS. 3A-3D for more detail). Since these packets are passed without regard to any of the information stored in database 26, management module 28 must compare the rules of database 26 to the information present in the packet header of each packet in order to determine whether the received packet should be stored.

MO

35-37

Those received packets which fulfill the rules of database 26 are then stored in a storage medium 30, which is preferably a high capacity digital data storage device such as a hard disk magnetic storage device, an optical disk, a CD-ROM, a ZIP or DVD drive, or a DAT cassette, or a combination of such devices according to the operational needs of specific applications, or any other suitable storage media. Preferably, the specific communication session or "telephone call", with which each data packet is associated, is also stored in order for that session to be reconstructed and displayed at a later time.

Playback

Upon request by the user, management module 28 can then retrieve one or more data packets from storage medium 30 which are associated with one or more communication sessions. The retrieved packet or packets are then transferred to a data restore module 32. Data restore module 32 is preferably capable of manipulating these retrieved packets to restore a particular communication session by using the RTP (Real Time Protocol). As described in further detail below with regard to FIGS. 4C and 5, in those systems which follow the RTP, the data packets are sent with a time stamp in the header rather than just a sequence number. Such a time stamp is necessary for audio and video stream data in order for the data packets to be reassembled such that the overall timing of the stream of data is maintained. Without such a time stamp, the proper timing would not be maintained, and the audio or video streams could not be accurately reconstructed.

49-65

The communication sessions are restored from the reconstructed streams of data packets by using the applicable

decoders

8/66-9/7

audio and/or video CODEC's. A CODEC is a non-linear method for the conversion of analog and digital data. Thus, an audio CODEC enables the digitized audio data in relevant data packets to be converted to analog audio data for display to the user as audible sounds, for example. Suitable CODEC's are described in greater detail below with regard to FIG. 5.

In order for the user to receive the display of the reconstructed communication session, system **10** preferably features an audio unit **34** and a video unit **36**, collectively referred to as a "communication session display unit". More preferably, both audio unit **34** and video unit **36** are capable of both receiving audio or video input, respectively, and of displaying audio or video output. At the very least, audio unit **34** and video unit **36** should be able to display audio or video output, respectively. For example, audio unit **34** could optionally include an microphone for input and a speaker or an earphone for output. Video unit **36** could optionally include a video monitor or display screen for output and a video camera for input, for example.

8-18

(34)

FIG. 2 is a schematic block diagram of system **10** of FIG. 1, showing the overall system of software modules of system **10** in more detail. Reference is also made, where appropriate, to flow charts showing the operation of these software modules in more detail (FIGS. 3A–3D and FIG. 5), as well as to descriptions of the headers of the different types of data packets (FIGS. 4A–4D).

As shown, system **10** again includes a connection to intranet **14** through NIC **16**. As the packets are transmitted through intranet **14**, NIC **16** intercepts these data packets and passes them to filtering module **24**.

Filtering module **24** has two components. A first filtering component **38** examines the header of the data packet, which should be an IP type packet with the correct header, as shown in FIG. 4A below. Next, first filtering component **38** passes the data packet to a second filtering component **40**. Second filtering component **40** then determines the type of IP data packet, which could be constructed according to the H.225, H.245, RTP or RTCP standards.

38/40

33-40

As shown with reference to FIG. 3A, first filtering component **38** and second filtering component **40** operate as follows. In step one, a packet is received by filtering module **24**. The packet is given to first filtering component **38**, which then determines whether the packet is an IP type packet in step two. Such a determination is performed according to the structure of the header of the data packet, an example of which is shown in FIG. 4A. A header **42** is shown as a plurality of boxes, each of which represents a portion or "field" of the header. The number of bytes occupied by each portion is also shown, it being understood that each layer consists of 32 bits. The first portion of the header, a "VERS" portion **44**, is the protocol version number. Next, an "H. LEN" portion **46** indicates the number of 32-bit quantities in the header. A "SERVICE TYPE" portion **48** indicates whether the sender prefers the datagram to travel over a route with minimal delay or a route with maximal throughput. A "TOTAL LENGTH" portion **50** indicates the total number of octets in both the header and the data.

41-48

MULTIPLE TO (40) ACCORDING TYPE (60)

In the next layer, an "IDENTIFICATION" portion **52** identifies the packet itself. A "FLAGS" portion **54** indicates whether the datagram is a fragment or a complete datagram. A "FRAGMENT OFFSET" portion **56** species the location of this fragment in the original datagram, if the datagram is fragmented. In the next layer, a "TIME TO LIVE" portion **58** contains a positive integer between 1 and 255, which is progressively decremented at each route traveled. When the value becomes 0, the packet will no longer be passed and is returned to the sender. A "TYPE" portion **60** indicates the type of data being passed. A "HEADER CHECKSUM" portion **62** enables the integrity of the packet to be checked by comparing the actual checksum to the value recorded in portion **62**.

The next layer of header **42** contains the source IP address **64**, after which the following layer contains the destination IP address **66**. An optional IP OPTIONS portion **68** is present, after which there is padding (if necessary) and a data portion **70** of the packet containing the data begins.

The structure of the header of the data packet is examined by first filtering component **38** to determine whether this header has the necessary data fields in the correct order, such that the header of the data packet has a structure according to header **42**. First filtering component **38** only allows those packets with the correct header structure to pass, as shown in step 3A. Otherwise, the packets are dumped as shown in step 3B. *

Those packets with the correct header, or "IP packets", are then passed to second filtering component **40**. Second filtering component **40** then performs the remainder of the filtering steps. In step 3A, second filtering component **40** examines the IP packets to determine their type from the data portion of the packet as shown in FIG. 4A. The packets could be in one of four categories: H.225, H.245, RTP and RTCP. The steps of the method for H.225 packets are shown in FIG. 3A, while the procedures for the remaining packet types are shown in FIGS. 3B–3D, respectively.

10/21-30

Once the type of the packet has been determined, both the packet itself and the information regarding the type of packet are both passed to management module **28**, as shown in FIG. 2. The packet is then passed to the relevant component within management module **28**, also as shown in FIG. 2, for the recording process to be performed. The recorded packets are stored in storage module **30**, as described in greater detail below with regard to FIGS. 3C and 3D.

10/21-

If the packet has been determined to be an H.225 packet according to the header of the packet (see FIG. 4B), the packet is passed to an H.225 call control module **78** within management module **28**, as shown in FIG. 2. The steps of the management method are as follows, with reference to FIG. 3A. In step 4A of FIG. 3A, the H.225 packet is examined to see if it is a setup packet, which is determined according to the structure of the data in the packet. This structure is specified in the H.225.0 recommendation, and includes at least the following types of information:

MULTIPLEXED TO DIFF DIRECT HANDLERS ACCORDING TO TYPE

protocolIdentifier (the version of H.225.0 which is supported);

h245Address (specific transport address on which H.245 signaling is to be established by the calling endpoint or gatekeeper);

sourceAddress (the H.323_ID's for the source);

sourceInfo (contains an EndpointType to enable the party being called to determine whether the call includes a gateway or not); and

destinationaddress (this is the address to which the endpoint wants to be connected).

Other types of data are also required, as specified in the H.225.0 Recommendation. This data structure enables H.225 call control module **78** to determine whether the packet is a setup packet.

If this packet is a setup packet, then the first branch of the method is followed. The source port is taken from a source

port field 74 of an H.225 header 72, and the destination port is taken from a destination port field 76 (see FIG. 4B). In step 5A, database 26 of FIG. 1 is then examined to determine whether either of the corresponding terminals is defined as a recording terminal; that is, whether communication sessions initiated by the IP address of this terminal should be monitored. If true, then in step 6A, the terminal status is set as a start session request from the terminal corresponding to the source port.

Alternatively, the packet is examined to see if it is a connect packet in step 4B, which is determined according to the structure of the data in the packet. This structure is specified in the H.225.0 recommendation, and includes at least the following types of information:

- protocolIdentifier (the version of H.225.0 which is supported);
- h245Address (specific transport address on which H.245 signaling is to be established by the calling endpoint or gatekeeper);
- destinationInfo (contains an EndpointType to enable the caller to determine whether the call includes a gateway or not); and
- conferenceID (contains a unique identifying number to identify the particular conference).

If the packet is a connect packet, then the second branch of the method is followed. In step 5B, the flag indicating the terminal status is examined to determine if the terminal status is set as a start session request. In step 6B, the details of the call signal are saved in a call progress database 78 of storage medium 30 (see FIG. 2). These details preferably include the source and destination IP addresses, the source and destination ports; the time at which the communication session was initiated, and any other relevant information. In step 7B, the status of the terminal is set to "wait for the logic channel".

If the packet has been determined to be an H.245 packet by second filtering component 40, the packet is passed to an H.245 call control module 82 within management module 28, as shown in FIG. 2. Such H.245 packets are necessary for H.245 signaling. H.245 signaling is established between two endpoints: an endpoint and a multi-point controller, or an endpoint and a Gatekeeper (see FIGS. 6 and 7 below for examples and a description of such endpoints). Each endpoint is capable of calling and of being called as part of a communication session. However, the system of the present invention only monitors, rather than initiating, such communication sessions. Thus, the system of the present invention uses the H.245 signaling to determine when the communication session has started in order to effectively record the necessary data packets for the storage and later reconstruction of the session.

35-50

The steps of the management method for H.245 packets are as follows, with reference to FIG. 3B. In step 1A of FIG. 3B, the H.245 packet is examined to determine if it is an open logical channel request packet. If it is, then in step 2A, the terminal status is examined to determine if the status is "wait for the logical channel". If so, then in step 3A the terminal status is set to "wait for acknowledgment".

Alternatively, the H.245 packet is examined to determine if it is an open logical channel acknowledgment packet, as shown in step 1B. If it is, then in step 2B, the terminal status is examined to determine if the status is "wait for acknowledgment". If so, then in step 3B the terminal status is set to "wait for terminal capability". In step 4B, the transport address of the "called" or destination terminal is saved. This transport address is taken from the destination port field 76 of header 72 (see FIG. 4B). It should be noted that H.225 and H.245 packets have identical header structures.

Also alternatively, the H.245 packet is examined to determine if it is a terminal capability set packet, as shown in step 1C. If it is, then in step 2C, the terminal capability is saved in call progress database 80 (see FIG. 2). In step 3C, the terminal status is set to "in call process", such that the communication session has been determined to be opened and such that management module 28 can now receive RTP data packets.

If the packet has been determined to be a RTP packet by second filtering component 40, the packet is passed to a RAS (registration, admissions and status) control module 84 within management module 28, as shown in FIG. 2. The steps of the management method for RTP packets are as follows, with reference to FIG. 3C. In step 1 of FIG. 3C, the terminal status is examined to see if it is "in call process". If so then in step 2, the RTP packets are saved in a RTP database 86 within storage medium 30 (see FIG. 2). FIG. 4C shows the structure of the RTP packet header, which can be used to identify the communication session from which the packet was taken.

Finally, if the packet has been determined to be a RTCP packet by second filtering component 40, the packet is passed to a RTCP control module 88 within management module 28, as shown in FIG. 2. The steps of the management method for RTCP packets are as follows, with reference to FIG. 3D. In step 1 of FIG. 3D, the terminal status is examined to see if it is "in call process". If so then in step 2, the RTCP packets are saved in call progress database 80 within storage medium 30 (see FIG. 2). FIG. 4D shows the structure of the RTCP packet header, which can be used to identify the communication session from which the packet was taken.

Thus, FIGS. 3A–3D illustrate the method of the present invention with regard to the filtering and storage of data packets which constitute the recorded communication session, as recorded by the system of the present invention as shown in FIGS. 1 and 2. Of course, in addition to recording such communication sessions, the system of the present invention is also able to retrieve and to replay these communication sessions to the user. The stored communication session, composed of stored data packets, can be retrieved and displayed by data restore unit 32 of FIG. 2, in conjunction with audio unit 34 and video unit 36. The method of retrieving and replaying sessions of interest is shown in FIG. 5, while certain other relevant portions of the system of the present invention are shown in FIG. 2.

In step 1 of FIG. 5, the user inputs the information concerning the communication session which is to be retrieved and replayed. This information preferably includes the terminal number, or other designation information concerning at least one of the parties of the communication session of interest; the time at which the session started; and the time at which the session ended. However, alternatively other information could be included in place of this information, as long as sufficient information is provided for the communication session of interest to be identified.

In step 2 of FIG. 5, call progress database 80 (see FIG. 2) is searched by data restore unit 32 in order to find the details of the communication session(s) in the specified time range. These details are then compared to the information entered by the user to locate at least one communication session of interest in the call range.

In step 3, RTP database 86 of storage medium 30 (see FIG. 2) is searched, again by data restore unit 32, to find substantially all data packets from the at least one communication session in the specified call range. Optionally and preferably, in step 4, if the audio portion communication

MULTIPLEX

session was recorded in stereo, then the data packets are divided into different audio channels.

In step 5, the data packets are restored by data restore unit 32 by an RTP (Real Time Protocol) software module 91 within data restore unit 32. RTP software module 91 orders the data packets within each channel according to the time stamp of each packet. As shown in FIG. 4C, an RTP packet header 92 features several important fields: a timestamp field 94, a synchronization source (SSRC) identifiers field 96 and a contributing source (CSRC) identifiers field 98. SSRC field 96 is used to determine the source of the RTP packets (the sender), which has a unique identifying address (the SSRC identifier). The CSRC identifier in CSRC field 98 is used in a conference with multiple parties, and indicates the SSRC identifier of all parties. Timestamp field 94 is used by RTP software module 91 to determine the relative time at which the data in each packet should be displayed.

For example, preferably the audio stream data of the audio speech of one person is synchronized to that person's lip movements as shown in the video stream, a process known as "lip synchronization". Such synchronization requires more than simply replaying audio and video data at certain relative time points, since the audio and video data packets may not arrive at the same time, and may therefore have slightly different timestamps.

Once the data packet has been correctly synchronized, the control of the display of the audio data is then performed by an audio component 102 of data restore unit 32 according to one or more audio CODEC's (see FIG. 2). The control of the display of the video data is then performed by a video component 104 of data restore unit 32 according to one or more video CODEC's (see FIG. 2).

Suitable CODEC's include, but are not limited to, an audio codec using CCITT Recommendation G.711(1988), Pulse Code Modulation (PCM) of voice frequencies; an audio codec using CCITT Recommendation G.722 (1988), 7 kHz audio-coding within 64 kbit/s; an audio codec using ITU-T Recommendation G.723.1 (1996), Speech coders: Dual rate speech coder for multimedia communications transmitting at 5.3. and 6.3 Kbps; an audio codec using CCITT Recommendation G.728 (1992), Coding of speech at 16 Kbps using low-delay code excited linear prediction; an audio codec using ITU-T Recommendation G.729 (1996), Coding of speech at 8 Kbps using conjugate structure algebraic code-excited linear-prediction (CS-ACELP); a video codec using ITU-T Recommendation H.261 (1993), Video codec for audiovisual services at px64 kbit/s; a video code using ITU-T Recommendation H.263 (1996), Video codingfor low bit rate communication; and substantially any other similar coding standard.

* 31-48

As shown in FIG. 2, the audio data is displayed by audio unit 34, which could include a loudspeaker, for example. The video data is displayed by video unit 36, which could include a display monitor screen, for example. Step 5 of FIG. 5 is then preferably repeated, such that substantially the entirety of the communication session is displayed. As shown in step 6, each data packet of the communication session is examined to see if the call time is over. If the individual session has not completed, preferably step 5 is repeated. Alternatively and preferably, if the call time is over, then call progress database 80 is searched to see if other communication sessions were recorded within the given time period, as shown in step 7. If there is at least one other such communication session, then preferably the method of FIG. 5 is repeated, starting from step 2.

* 49-50

Repeat

According to preferred embodiments of the present invention, several configurations of the computer logging system are possible, examples of which are shown in FIGS. 6 and 7.

According to a first embodiment of the system of the present invention, shown in FIG. 6, a typical basic configuration system 104 includes a single communication session management unit 13, substantially as shown in FIGS. 1 and 2, according to the present invention. Communication session management unit 13 manages communication in a stand-alone intranet such as a LAN 106. LAN 106 is connected both to communication session management unit 13 and to a plurality of terminals 108, designated as "T1", "T2" and so forth, which follow the H.323 protocol. Each terminal 108 is an endpoint on LAN 106 which provides for real-time, two-way communications with another terminal 108, a gateway 110, or a multipoint control unit 112. This communication consists of control, indications, audio streams, video streams, and/or data. Terminal 108 is optionally only capable of providing such communication for audio only, audio and data, audio and video, or audio, data and video. As noted previously in the "Description of the Background Art" section, the H.323 entity could be a terminal which is capable of providing audio and/or video communication as a "LAN telephone", but could also be a stand-alone audio or video telephone.

Gateway 110 (GW) is constructed according to H.323 and is an endpoint on LAN 106 which provides for real-time, two-way communications between terminals 108 on LAN 106 and other suitable terminals on a WAN (not shown), or to another such Gateway (not shown). Other suitable terminals include those complying with Recommendations H.310 (H.320 on B-ISDN), H.320 (ISDN), H.321 (ATM), H.322 (GQOS-LAN), H.324 (GSTN), H.324M (Mobile), and V.70 (DSVD).

Multipoint Control Unit (MCU) 112 is an endpoint on LAN 106 which enables three or more terminals 108 and gateways 110 to participate in a multipoint conference. *

Preferably, system 104 also features a gatekeeper (GK) 114, which is an H.323 entity on LAN 106 which provides address translation and controls access to LAN 106 for terminals 108, gateways 110 and MCUs 112. Gatekeeper 114 may also provide other services to terminals 108, gateways 110 and MCUs 112 such as bandwidth management and locating gateways 110. Preferably, gatekeeper 114 enables the IP address of terminals 108 on LAN 106 to be determined, such that the correct IP address can be determined "on the fly".

In addition, LAN 106 may support non audio visual devices for regular T.120 data applications such as electronic whiteboards, still image transfer, file exchange, database access, etc.

In basic system 104, a single, stand-alone communication session management unit 13 is used for monitoring, logging and retrieval of all audio and/or visual calls either between any two or more terminals 108 attached to LAN 106 or any call to which one or more of these terminals 108 is a party. 49-54 However, for the preferred embodiment of the system of FIG. 6 which includes gatekeeper 114, as well as for the system of FIG. 7, once the communication session has been opened, preferably RAS control module 84 also performs RAS signaling between the management control module and NIC 16 where necessary for the configuration of the system. Such signaling uses H.225.0 messages to perform registration issions, bandwidth changes, status, and disengage procedures between endpoints and gatekeepers. These messages are passed on a RAS Signaling Channel, which is independent from the Call Signaling Channel and the H.245 Control Channel. H.245 open logical channel procedures are not used to establish the RAS Signaling Channel. In LAN environments which contain a Gatekeeper (a Zone), the

RAS Signaling Channel is opened between the endpoint and the Gatekeeper. The RAS Signaling Channel is opened prior to the establishment of any other channels between H.323 endpoints.

FIG. 7 shows a second embodiment of the system of the present invention as a zone configuration system 116. A zone 118 is the collection of all terminals (Tx) 108, gateways (GW) 110, and Multipoint Control Units (MCU) 112 managed by a single gatekeeper (GK) 114. Zone 118 includes at least one terminal 108, but does not necessarily include one or more gateways 110 or MCUs 112. Zone 118 has only one gatekeeper 114 as shown. However, in the preferred embodiment shown, zone 118 is preferably independent of LAN topology and preferably includes multiple LAN segments 120 which are connected using routers (R) 122 as shown or other similar devices.

Each monitored LAN segment 120 has a local communication management unit 124 according to the present invention, of which two are shown. A central management unit 126 according to the present invention controls all local communication management units 124. In addition to centralized database and control services, central management unit 126 can be used for the real-time monitoring and off-line restoration of audio and/or video communication sessions from a single point. Central management unit 126 is optionally and preferably either a dedicated unit similar in structure to local communication management units 124 but without the storage capability, or central management unit 126 is alternatively and preferably integrated with local communication management units 124 to provide the functionality of both local communication management unit 124 and central management unit 126 in a single station. Local communication management units 124 are preferably either communication management units 13 substantially as described in FIGS. 1 and 2, or alternatively and preferably are simpler units which lack the capability to retrieve and display a communication session locally.

In still another preferred embodiment of the present invention (not shown), multi-user operation based on Client/Server architecture is preferably supported for basic system 104 and zone system 116. An unlimited number of "Client" stations may be connected anywhere on the LAN, providing users with management and monitoring/retrieval capabilities determined by the authorization level of each specific user.

It will be appreciated that the above descriptions are intended only to serve as examples, and that many other embodiments are possible within the spirit and the scope of the present invention.

What is claimed is:

1. A system for managing a computer network-based telephone session over a computer network, the computer network being divided into a plurality of segments, the system comprising:

(a) a network connector for connecting to the computer network and for receiving data packets from a single segment of the computer network;

(b) a filtering unit for filtering said data packets from said single segment and for accepting said data packets substantially only if said data packets contain data selected from the group consisting of audio data and video data, such that said data packets form at least a part of the computer network-based telephone session and such that said data packets are selected data packets;

(c) a management unit for receiving said selected data packets from said single segment and for storing said selected data packets, such that said selected data packets are stored data packets, wherein said management unit and said filtering unit form a local management unit for said single segment of the computer network, said local management unit analyzing said selected data packets if the computer network-based telephone session occurs within said single segment;

(d) a storage medium for receiving and storing said stored data packets from said local management unit, such that at least a portion of the computer network-based telephone session is stored; and

(e) a central management unit for controlling each local management unit, said central management unit controlling storage in said storage medium and said central management unit analyzing said selected data packets from the computer network-based telephone session if the computer network-based telephone session includes data packets transmitted on a plurality of segments of the computer network.

2. The system of claim 1, further comprising:

(f) a data restore unit for retrieving and displaying said at least a portion of the computer network-based telephone session, said data restore unit requesting said data packets from said storage medium through said central management unit, and said data restore unit reconstructing said data packets for displaying said at least a portion of the computer network-based telephone session.

3. The system of claim 2, wherein said data restore unit further comprises a communication session display unit for displaying at least a portion of the computer network-based telephone session.

4. The system of claim 3, wherein said communication session display unit is selected from the group consisting of a video unit and an audio unit.

5. The system of claim 2, further comprising:

(g) a database connected to said filtering unit for storing filtering information, said filtering information including at least one IP address of a party whose computer network-based telephone sessions are monitored;

wherein said filtering unit accepts said data packets according to said filtering information, such that said filtering unit substantially only accepts said data packets if said data packets fulfill said filtering information.

6. The system of claim 5, further comprising:

(g) a user computer for receiving at least one command of a user and for displaying information to said user, such that said user determines said filtering information according to said at least one command of said user.

7. The system of claim 6, wherein the computer network is selected from the group consisting of a LAN (local area network) and a WAN (wide area network).

8. The system of claim 7, wherein the computer network is a LAN (local area network).

9. The system of claim 1, wherein said network connector is a network interface card.

10. A method for storing at least a portion of a computer network-based telephone session performed on a computer network, the computer network-based telephone session being performed between a packet source and a packet destination, the steps of the method being performed by a data processor, the method comprising the steps of:

(a) receiving a data packet from the packet source on the computer network;

(b) analyzing said data packet to determine if said data packet is a computer network-based telephone session packet;

(c) if said data packet is said computer network-based telephone session packet, filtering at least data in said data packet to determine if said data includes computer network-based telephone session data;

(d) if said data includes computer network-based telephone session data, analyzing said computer network-based telephone session data; and

(e)storing said computer network-based telephone session packet to form a stored packet according to said type, such that said stored data packet forms at least a portion of the computer network-based telephone session.

11. The method of claim **10**, wherein said data packet has a header and the step of analyzing said data packet in step (d) further comprises the step of:

(i) filtering said header of said data packet to retrieve header data related to the computer network-based telephone session.

12. The method of claim **11**, wherein substep (i) of step (d) further comprises the step of:

(1) analyzing said header data to determine if said data packet is an IP packet.

13. The method of claim **12**, wherein the step of analyzing said header data in substep (1) further comprises the steps of:

(i) examining said header of said IP packet to determine an IP address of said packet source;

(ii) determining if said IP address is a recorded IP address;

(iii) passing said IP packet to form a passed IP packet substantially only if said IP address is said recorded IP address; and

(iv) alternatively, dumping said IP packet.

14. The method of claim **13**, wherein the step of determining if said IP address is said recorded IP address is performed by comparing said IP address to a list of IP addresses from packet sources, such that if said IP address is included in said list, said IP address is said recorded IP address.

15. The method of claim **13**, wherein step (d) further comprises the steps of:

(ii) analyzing said IP packet to determine whether said passed IP packet is an H.225 packet, a H.245 packet, an RTP packet or an RTCP packet;

(iii) if said type of said passed IP packet is said H.225 packet, determining whether said H.225 packet is a setup packet or a connect packet;

(iv) if said H.225 packet is said setup packet, setting a status flag as "start session request";

(v) alternatively, if said H.225 packet is said connect packet and said status flag is "start session request", storing at least one detail of the computer network-based telephone session; and

(vi) setting said status flag as "wait for logic channel".

16. The method of claim **15**, wherein step (d) further comprises the steps of:

(vii) alternatively, if said type of said passed IP packet is said H.245 packet, determining whether H.245 packet is an open logical channel request packet, an open logical channel acknowledgment packet or a terminal capability set packet;

(viii) if said H.245 packet is said open logical channel request packet and said status flag is "wait for logic channel", setting said status flag as "wait for acknowledgment";

(xi) alternatively, if said H.245 packet is said open logical chanel acknowledgment packet and said status flag is "wait for acknowledgment", performing the steps of:

(A) setting said status flag as "wait for terminal capability"; and

(B) saving a transport address of the destination of the communication session; and

(xii) also alternatively, if said H.245 packet is said terminal capability set packet, performing the steps of:

(A) storing a capability of the packet destination from said terminal capability packet; and

(B) setting said status flag as "in call progress".

17. The method of claim **16**, wherein if said status flag is "in call process" and said type of said passed IP packet is said RTP packet, storing said RTP packet.

18. The method of claim **16**, wherein if said status flag is "in call process" and said type of said passed IP packet is said RTCP packet, storing said RTCP packet.

19. The method of claim **10**, further comprising the steps of:

(f) retrieving said stored data packet to form a retrieved data packet; and

(g) reconstructing at least a portion of the computer network-based telephone session according to said retrieved data packet.

20. The method of claim **19**, wherein the step of retrieving said data packet of step (f) includes the steps of:

(i) retrieving a source IP address of the packet source, a start time of the network-based telephone session, and an end time of the computer network-based telephone session; and

(ii) selecting at least one computer network-based telephone session according to said source IP address, said start time and said end time.

21. The method of claim **19**, wherein the step of reconstructing at least a portion of the computer network-based telephone session of step (g) includes displaying audio data.

22. The method of claim **19**, wherein the step of reconstructing at least a portion of the computer network-based telephone session of step (g) includes displaying video data.

23. The method of claim **19**, wherein the step of reconstructing at least a portion of the computer network-based telephone session of step (g) further comprises the steps of:

(i) receiving substantially only RTP packets;

(ii) examining a header of said RTP packets to determine a time stamp for each of said RTP packets; and

(iii) displaying said RTP packets in order according to said time stamp.

* * * * *

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US 6130880 A	23	0	0
US 6122665 A	18	0	0
Total	41	0	0

US006130880A

United States Patent [19]

Naudus et al.

[11] **Patent Number: 6,130,880**

[45] **Date of Patent: Oct. 10, 2000**

[54] **METHOD AND APPARATUS FOR ADAPTIVE PRIORITIZATION OF MULTIPLE INFORMATION TYPES IN HIGHLY CONGESTED COMMUNICATION DEVICES**

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[73] Assignee: **3Com Corporation**, Santa Clara, Calif.

[21] Appl. No.: **09/044,958**

[22] Filed: **Mar. 20, 1998**

[51] **Int. Cl.**^{7} **G01R 31/08**; H04J 3/16; H04J 3/22

[52] **U.S. Cl.** **370/235**; 370/468

[58] **Field of Search** 370/235, 237, 370/230, 231, 229, 400, 412, 401, 466, 468, 465, 477

[56] **References Cited**

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Primary Examiner—Alpus H. Hsu
Assistant Examiner—Duc Ho
Attorney, Agent, or Firm—McDonnell Boehnen Hulbert & Berghoff; Stephen Lesavich

[57] **ABSTRACT**

A method and apparatus are used in a gateway to discard selected frames received with a selected encoded-information-type from a communication link with a larger bandwidth to avoid overflowing an internal delay variance removing queue used for protocol translation to a communication link with a smaller bandwidth. The discarded frames do not decrease the quality of translated information. A visual delay variance removing queue congestion indicator is included to indicate three levels of congestion in the delay variance removing queue for received frames. The method and apparatus are used in a multimedia gateway which is translating audio/video conferencing protocols (e.g., H.320, H.323/LAN H.323/PPP and H.324) received from a communication link with a large bandwidth and sent to a communication link with a smaller bandwidth.

19 Claims, 13 Drawing Sheets

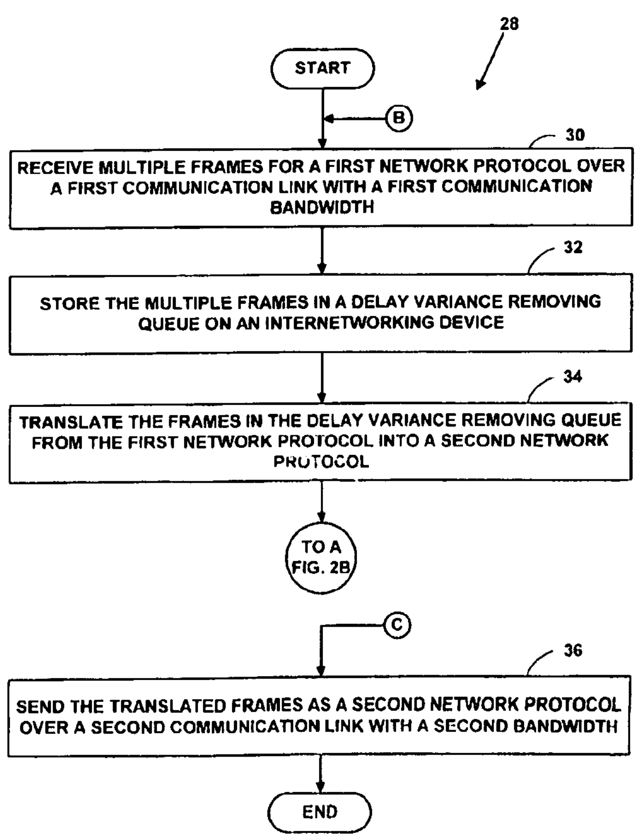


FIG. 1

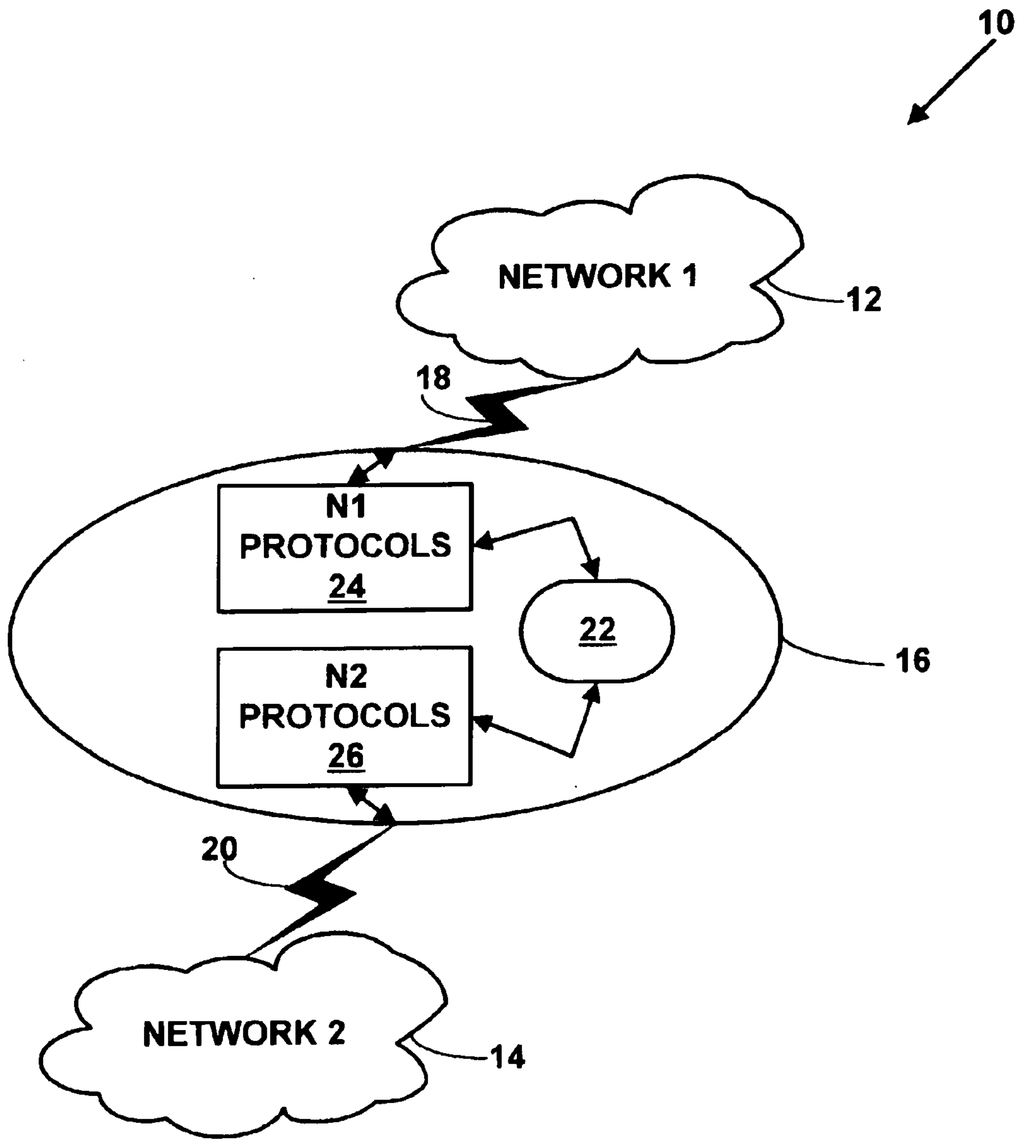

10
NETWORK 1
12
18
N1
PROTOCOLS
24
22
16
N2
PROTOCOLS
26
20
NETWORK 2
14

FIG. 2A

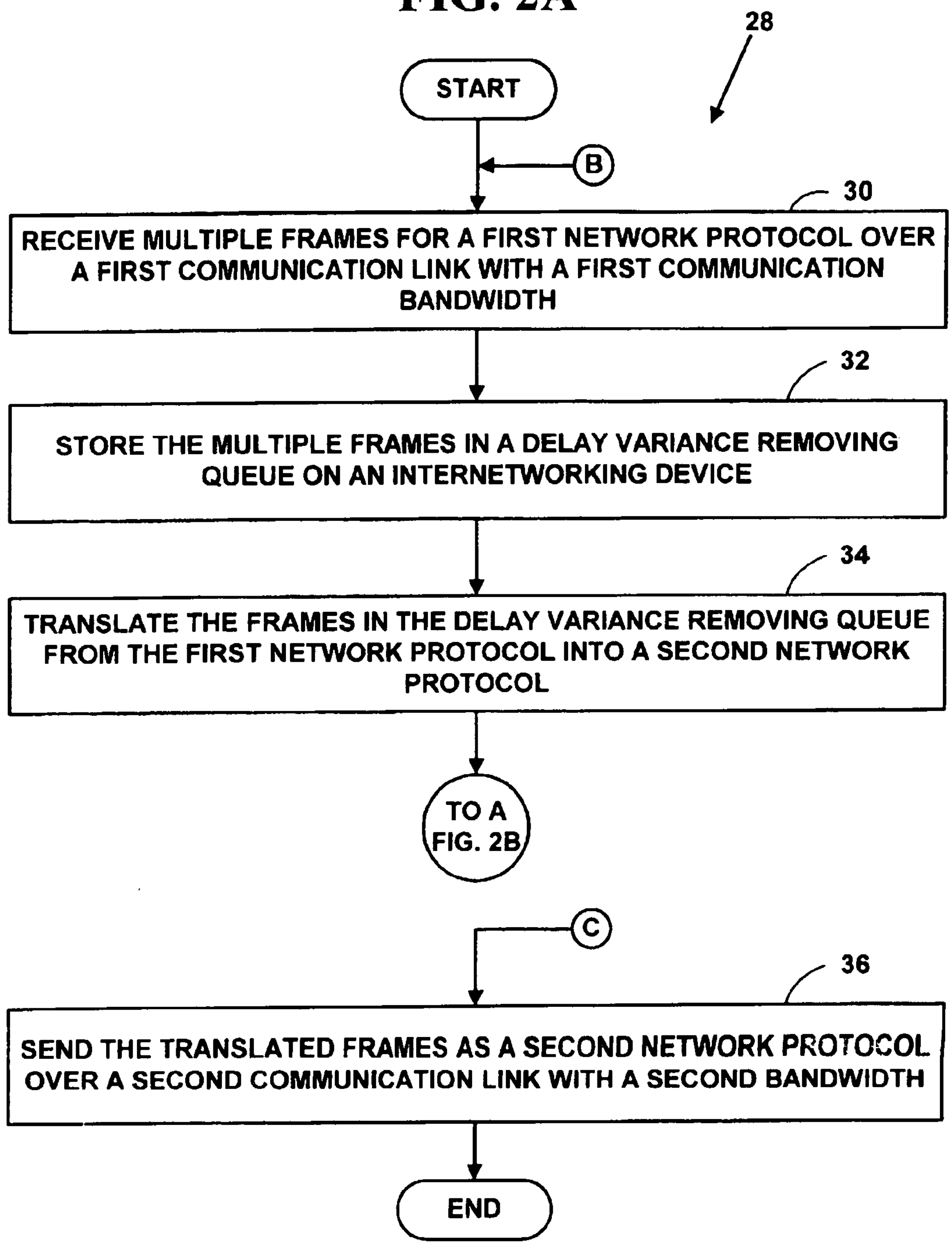

28
START
B
30
RECEIVE MULTIPLE FRAMES FOR A FIRST NETWORK PROTOCOL OVER A FIRST COMMUNICATION LINK WITH A FIRST COMMUNICATION BANDWIDTH
32
STORE THE MULTIPLE FRAMES IN A DELAY VARIANCE REMOVING QUEUE ON AN INTERNETWORKING DEVICE
34
TRANSLATE THE FRAMES IN THE DELAY VARIANCE REMOVING QUEUE FROM THE FIRST NETWORK PROTOCOL INTO A SECOND NETWORK PROTOCOL
TO A FIG. 2B
C
36
SEND THE TRANSLATED FRAMES AS A SECOND NETWORK PROTOCOL OVER A SECOND COMMUNICATION LINK WITH A SECOND BANDWIDTH
END

FIG. 2B

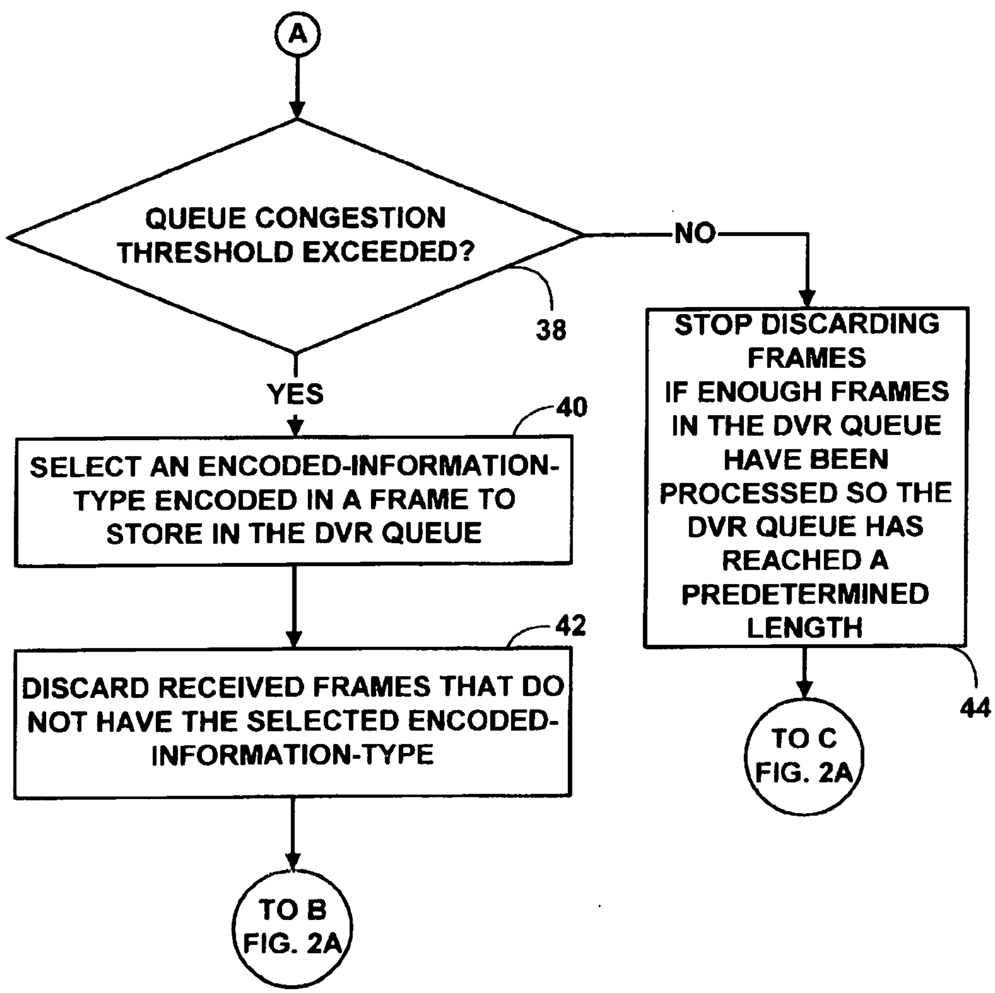

A
QUEUE CONGESTION THRESHOLD EXCEEDED?
38
NO
YES
STOP DISCARDING FRAMES IF ENOUGH FRAMES IN THE DVR QUEUE HAVE BEEN PROCESSED SO THE DVR QUEUE HAS REACHED A PREDETERMINED LENGTH
40
SELECT AN ENCODED-INFORMATION-TYPE ENCODED IN A FRAME TO STORE IN THE DVR QUEUE
42
DISCARD RECEIVED FRAMES THAT DO NOT HAVE THE SELECTED ENCODED-INFORMATION-TYPE
44
TO C FIG. 2A
TO B FIG. 2A

FIG. 3

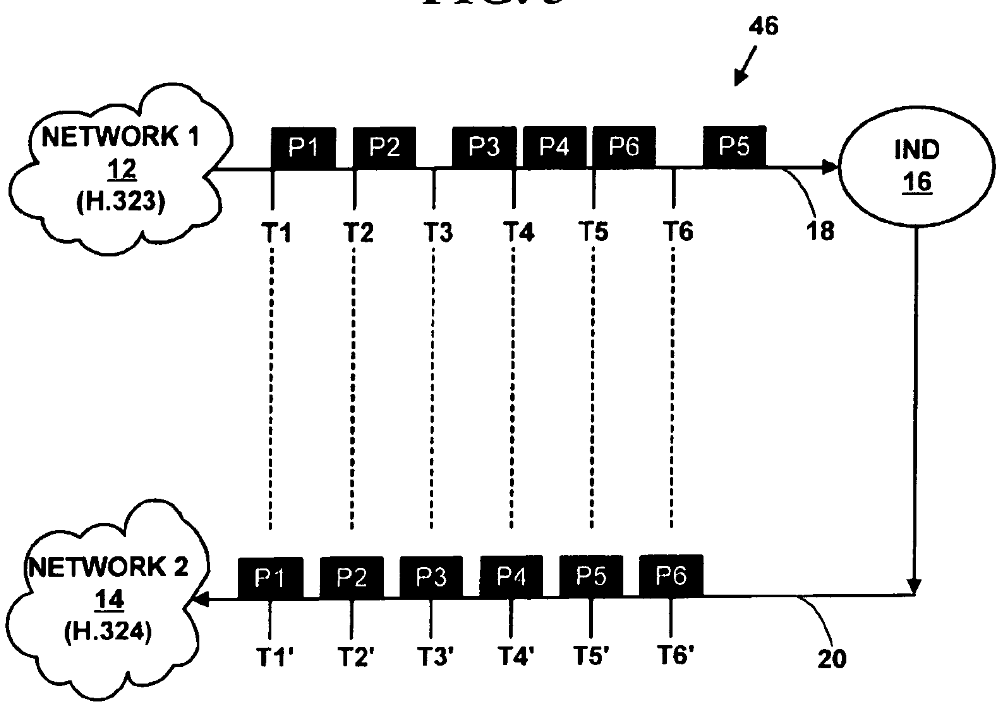

46
NETWORK 1
12
(H.323)
P1
P2
P3
P4
P6
P5
IND
16
T1
T2
T3
T4
T5
T6
18
NETWORK 2
14
(H.324)
P1
P2
P3
P4
P5
P6
T1'
T2'
T3'
T4'
T5'
T6'
20

FIG. 4

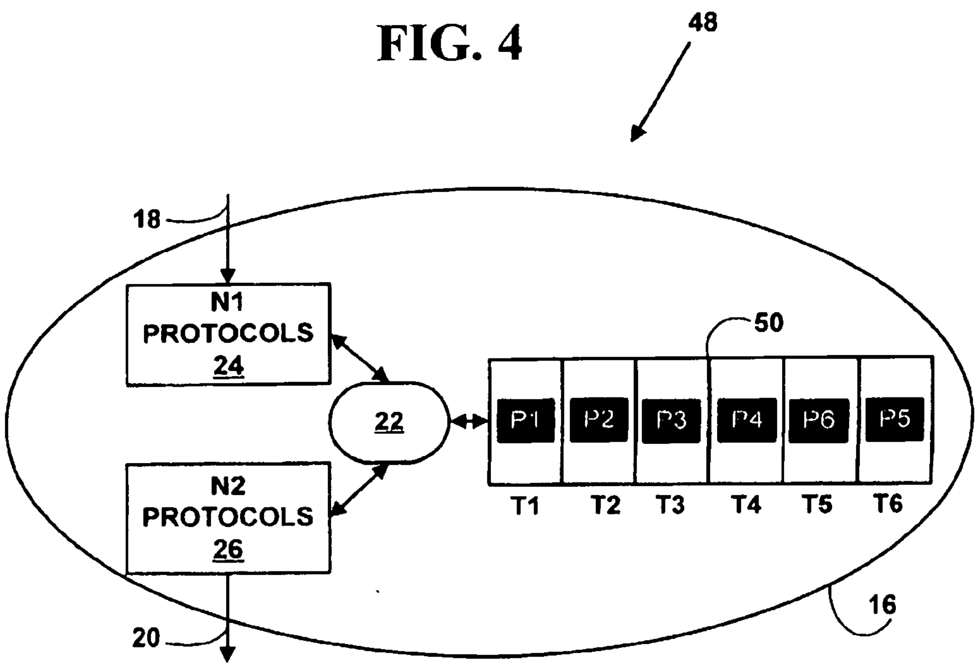

48
18
N1
PROTOCOLS
24
22
50
P1
P2
P3
P4
P6
P5
T1
T2
T3
T4
T5
T6
N2
PROTOCOLS
26
20
16

FIG. 5

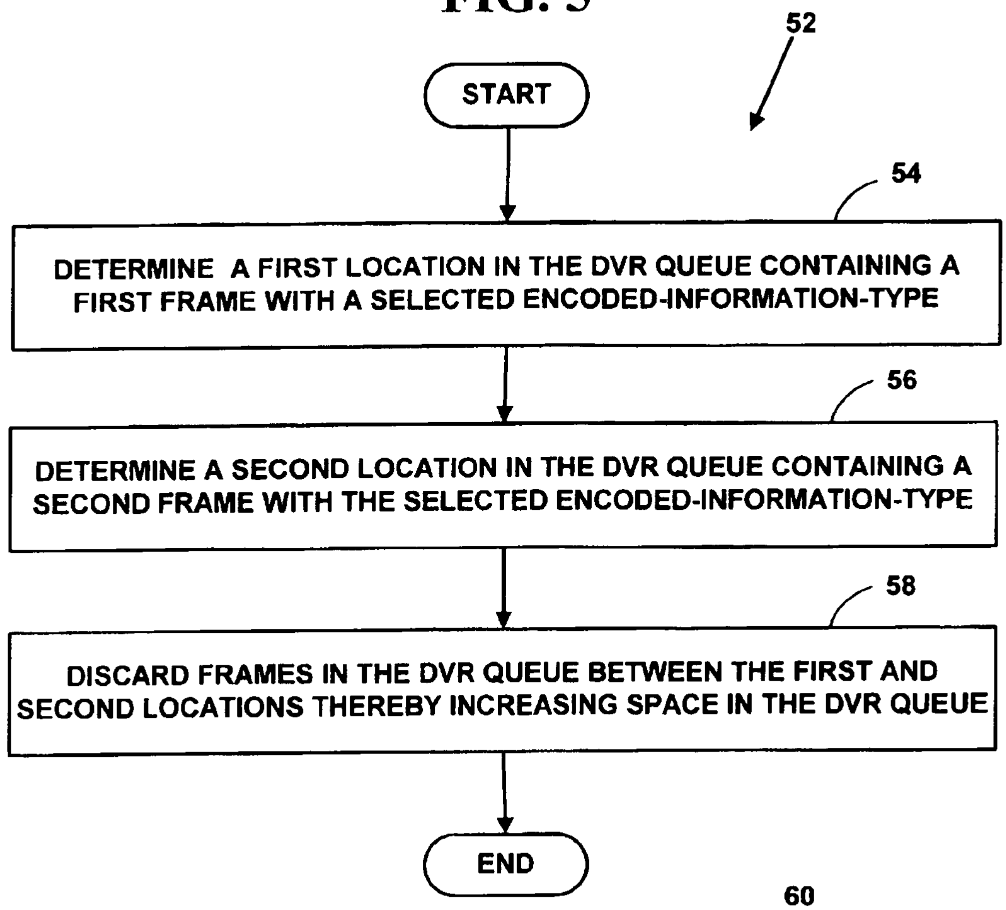

52
START
54
DETERMINE A FIRST LOCATION IN THE DVR QUEUE CONTAINING A FIRST FRAME WITH A SELECTED ENCODED-INFORMATION-TYPE
56
DETERMINE A SECOND LOCATION IN THE DVR QUEUE CONTAINING A SECOND FRAME WITH THE SELECTED ENCODED-INFORMATION-TYPE
58
DISCARD FRAMES IN THE DVR QUEUE BETWEEN THE FIRST AND SECOND LOCATIONS THEREBY INCREASING SPACE IN THE DVR QUEUE
END

FIG. 6

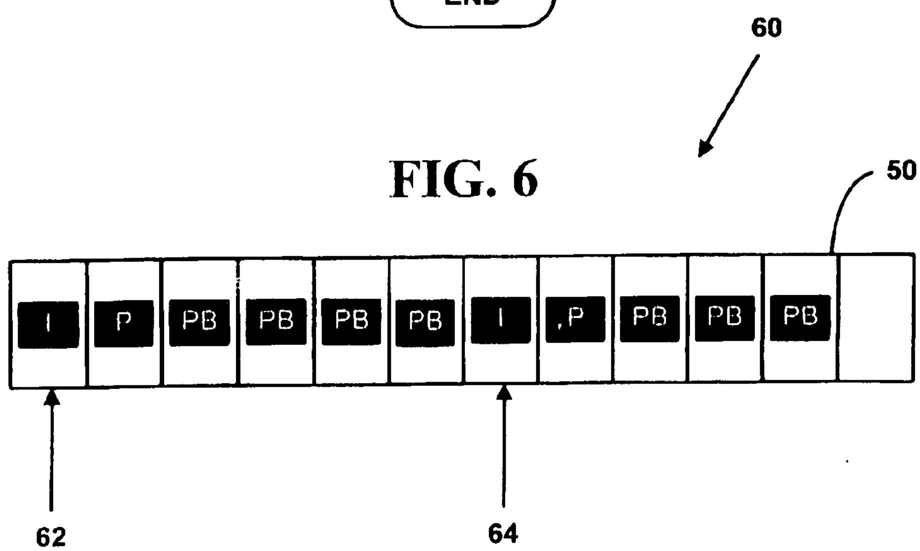

60
50
I
P
PB
PB
PB
PB
I
P
PB
PB
PB
62
64

FIG. 7

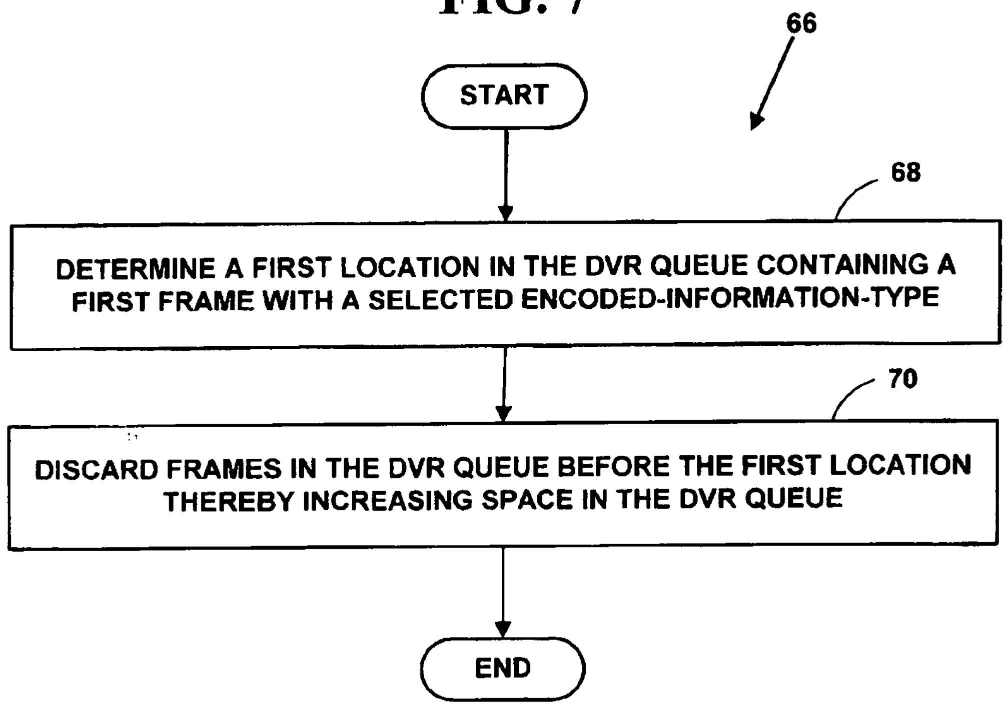

66
START
68
DETERMINE A FIRST LOCATION IN THE DVR QUEUE CONTAINING A FIRST FRAME WITH A SELECTED ENCODED-INFORMATION-TYPE
70
DISCARD FRAMES IN THE DVR QUEUE BEFORE THE FIRST LOCATION THEREBY INCREASING SPACE IN THE DVR QUEUE
END

FIG. 8

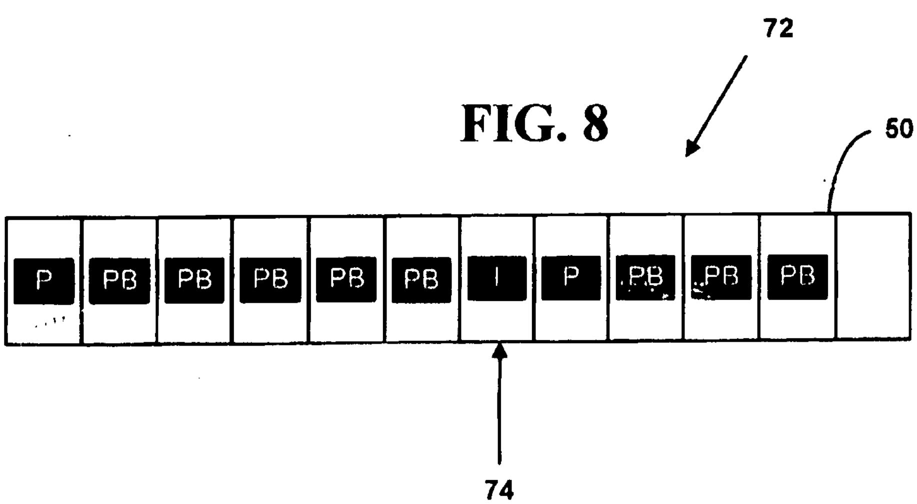

72
50
P
PB
PB
PB
PB
PB
I
P
PB
PB
PB
74

FIG. 9

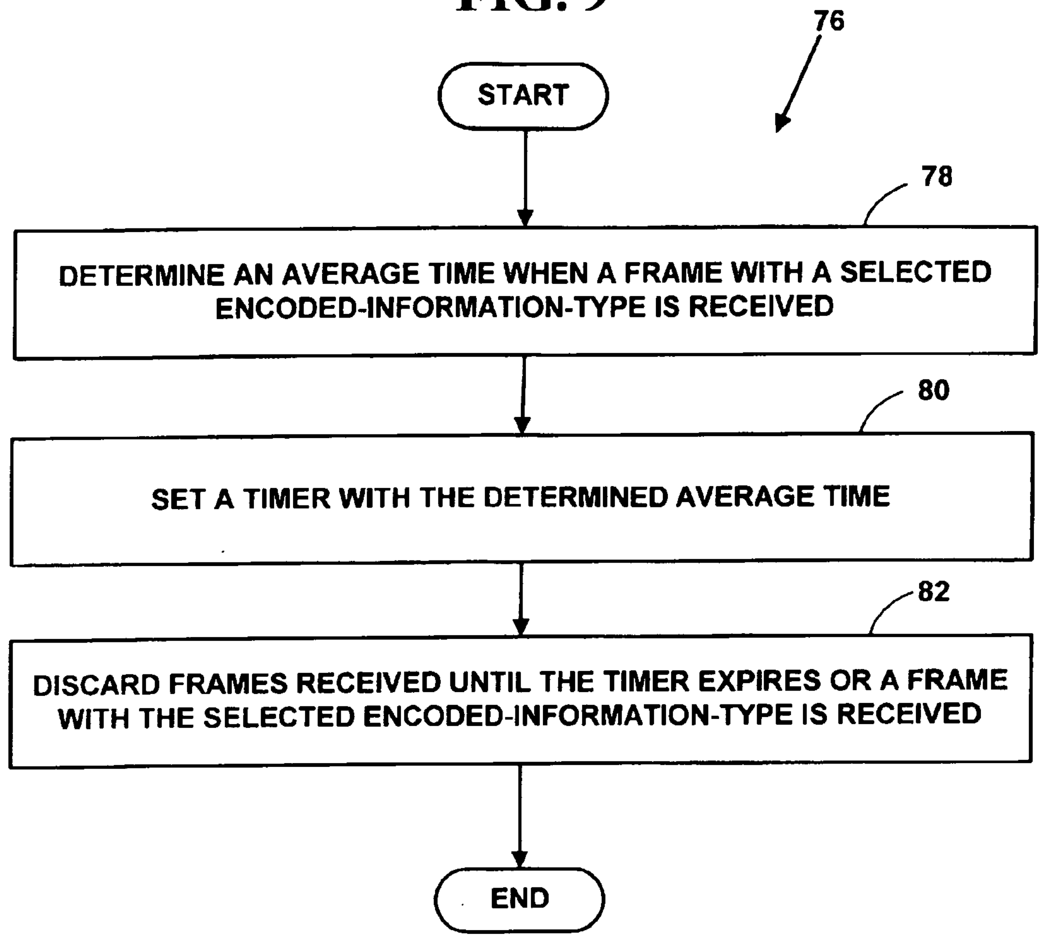

76
START
78
DETERMINE AN AVERAGE TIME WHEN A FRAME WITH A SELECTED ENCODED-INFORMATION-TYPE IS RECEIVED
80
SET A TIMER WITH THE DETERMINED AVERAGE TIME
82
DISCARD FRAMES RECEIVED UNTIL THE TIMER EXPIRES OR A FRAME WITH THE SELECTED ENCODED-INFORMATION-TYPE IS RECEIVED
END

FIG. 10

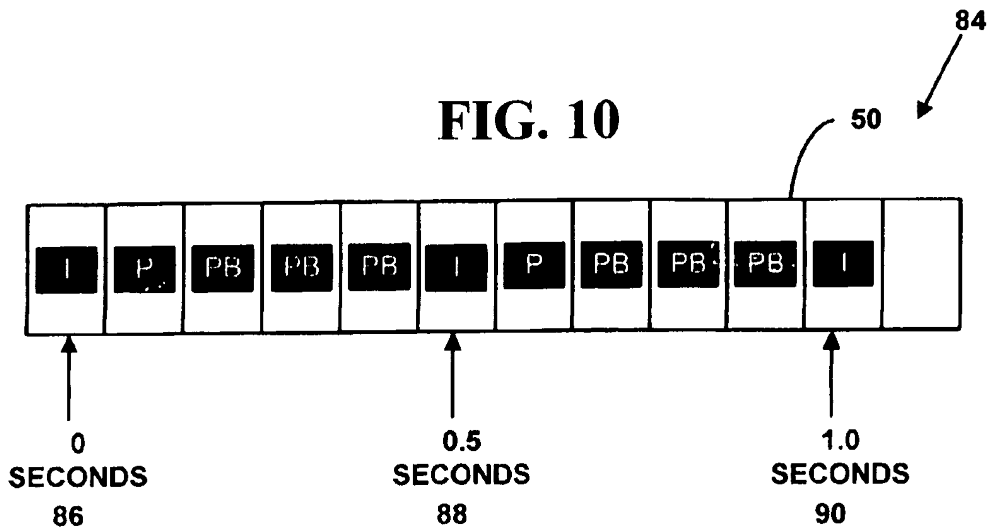

84
50
I
P
PB
PB
PB
I
P
PB
PB
PB
I
0 SECONDS
86
0.5 SECONDS
88
1.0 SECONDS
90

FIG. 11

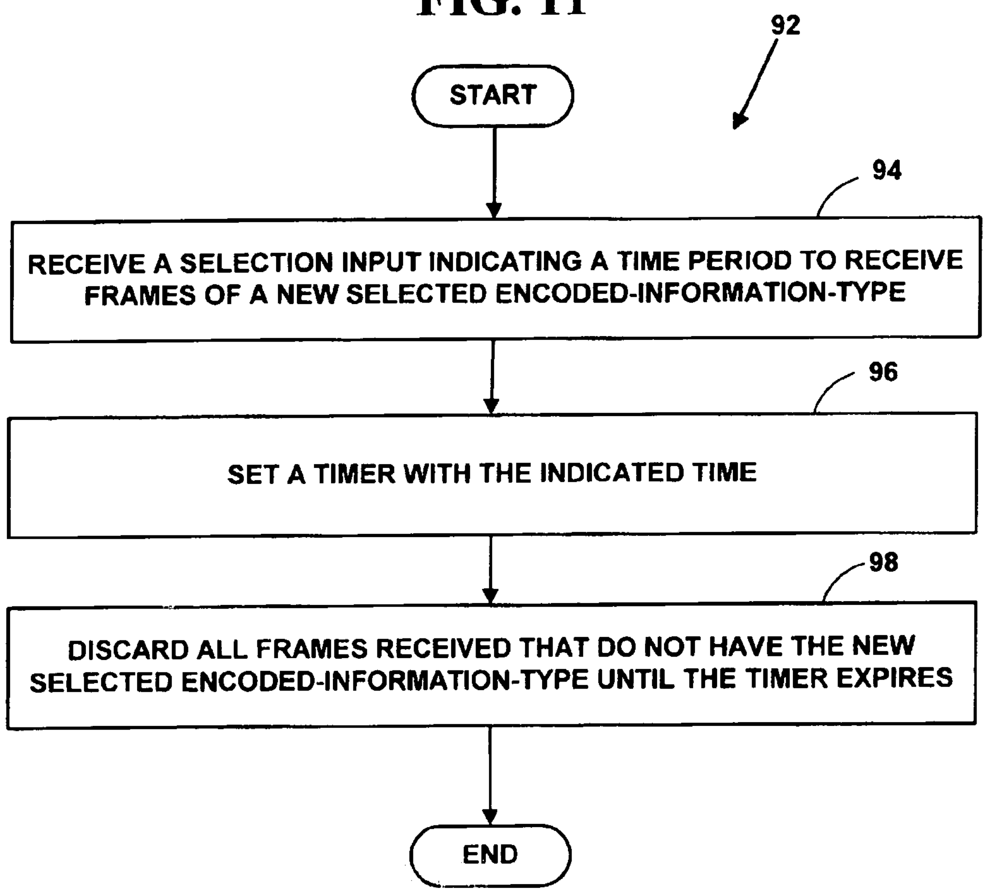

92
START
94
RECEIVE A SELECTION INPUT INDICATING A TIME PERIOD TO RECEIVE FRAMES OF A NEW SELECTED ENCODED-INFORMATION-TYPE
96
SET A TIMER WITH THE INDICATED TIME
98
DISCARD ALL FRAMES RECEIVED THAT DO NOT HAVE THE NEW SELECTED ENCODED-INFORMATION-TYPE UNTIL THE TIMER EXPIRES
END

FIG. 12A

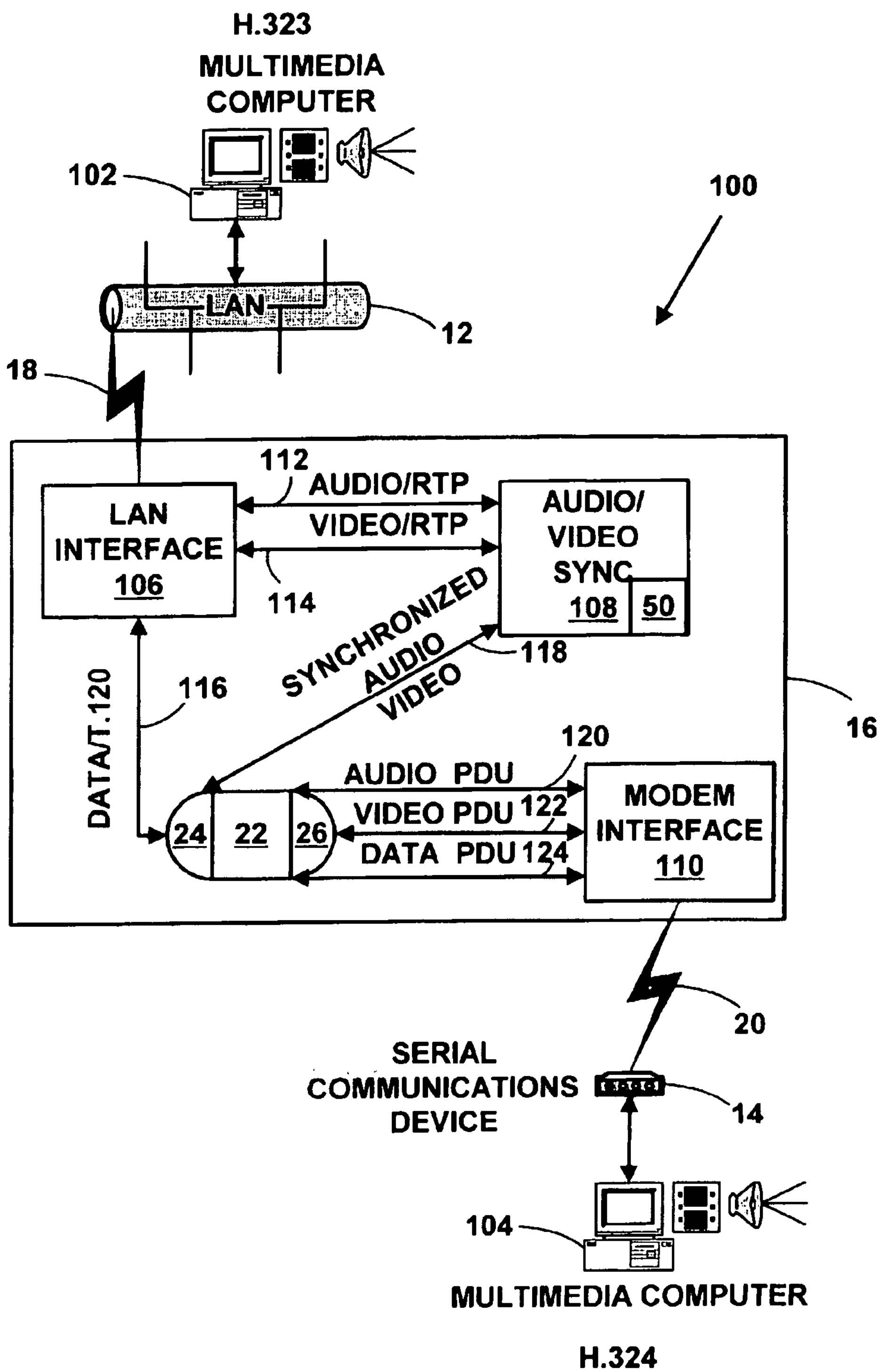

H.323
MULTIMEDIA
COMPUTER
102
100
LAN
12
18
112
AUDIO/RTP
LAN
INTERFACE
106
VIDEO/RTP
AUDIO/
VIDEO
SYNC
108
50
114
SYNCHRONIZED
AUDIO
VIDEO
118
DATA/T.120
116
16
120
AUDIO PDU
VIDEO PDU 122
DATA PDU 124
24
22
26
MODEM
INTERFACE
110
20
SERIAL
COMMUNICATIONS
DEVICE
14
104
MULTIMEDIA COMPUTER
H.324

FIG. 12B

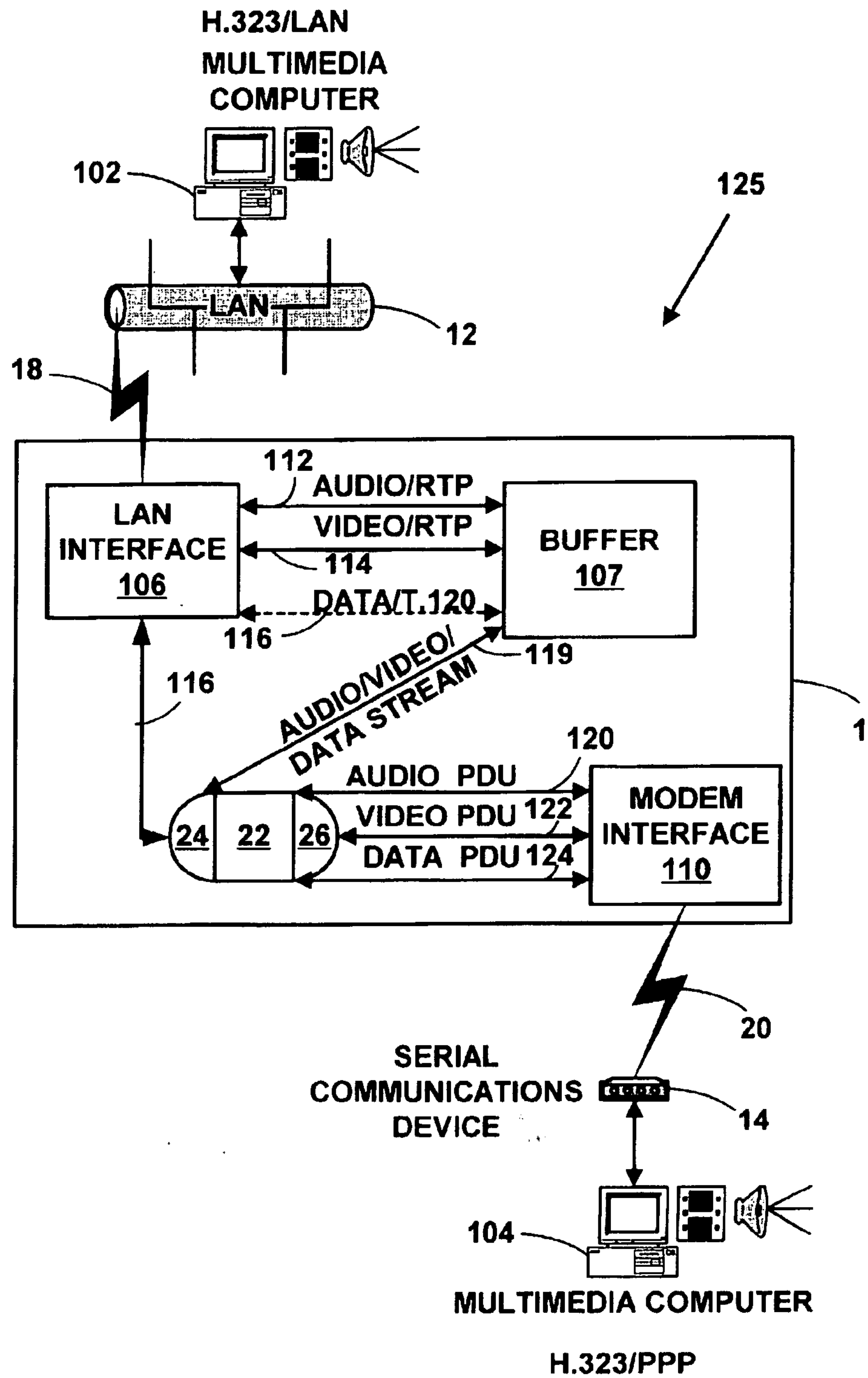

H.323/LAN
MULTIMEDIA
COMPUTER
102
125
LAN
12
18
112
AUDIO/RTP
LAN
INTERFACE
106
VIDEO/RTP
BUFFER
107
114
DATA/T.120
116
119
AUDIO/VIDEO/
DATA STREAM
116
16
120
AUDIO PDU
MODEM
INTERFACE
110
24
22
26
VIDEO PDU
122
DATA PDU
124
20
SERIAL
COMMUNICATIONS
DEVICE
14
104
MULTIMEDIA COMPUTER
H.323/PPP

FIG. 13

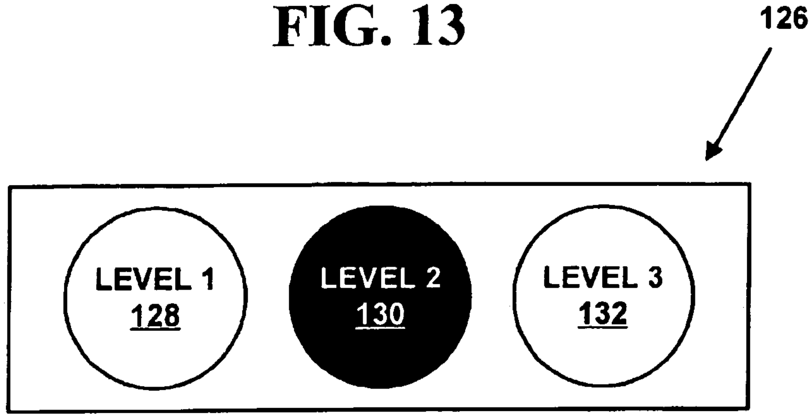


FIG. 14A

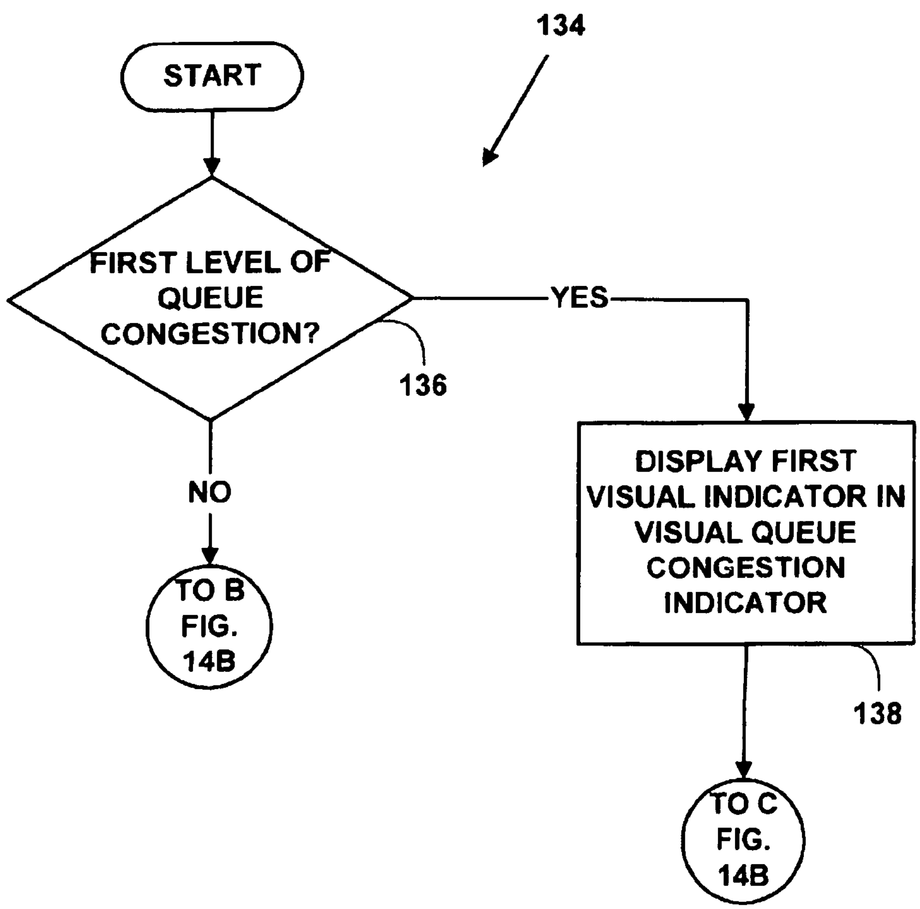


FIG. 14B

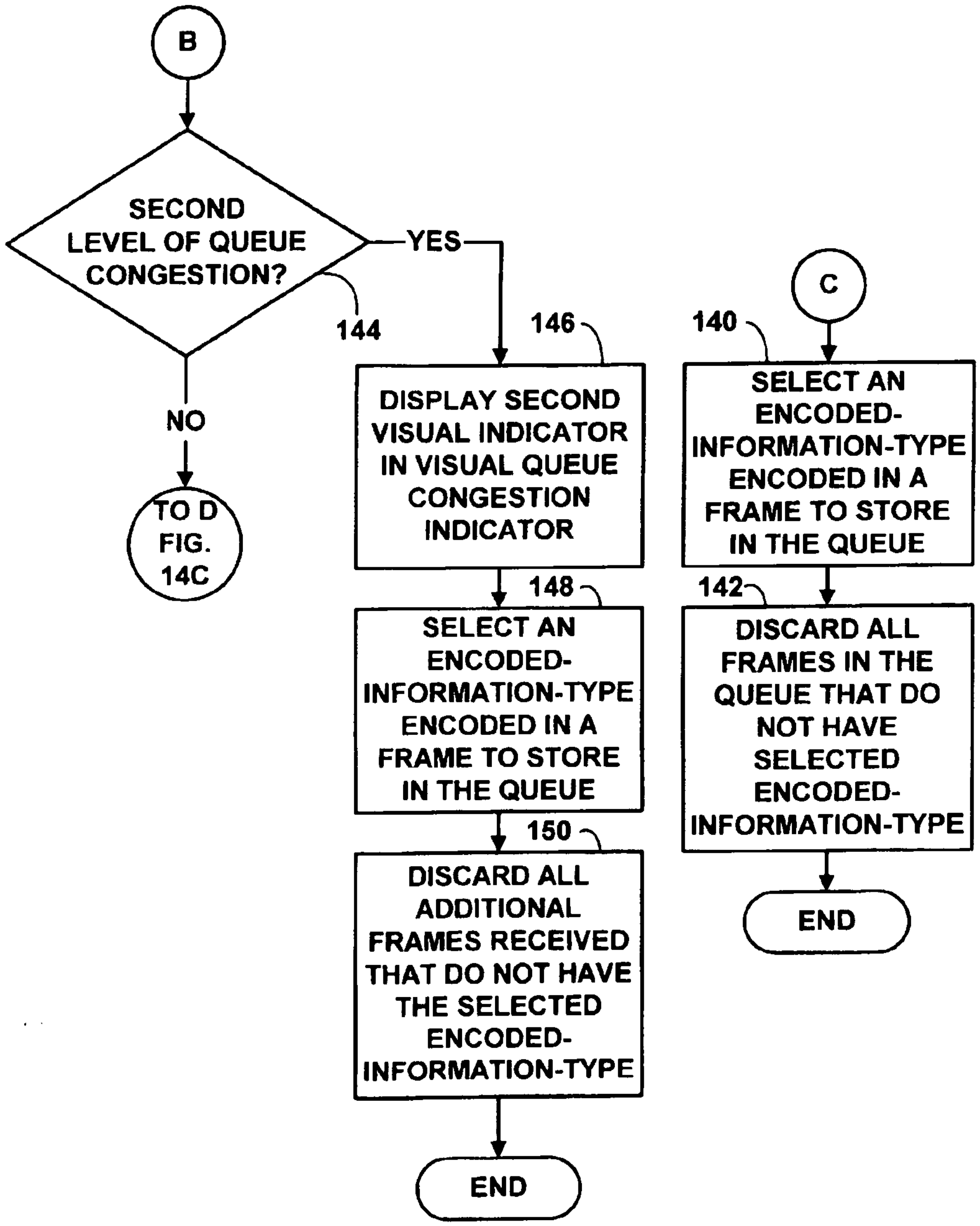
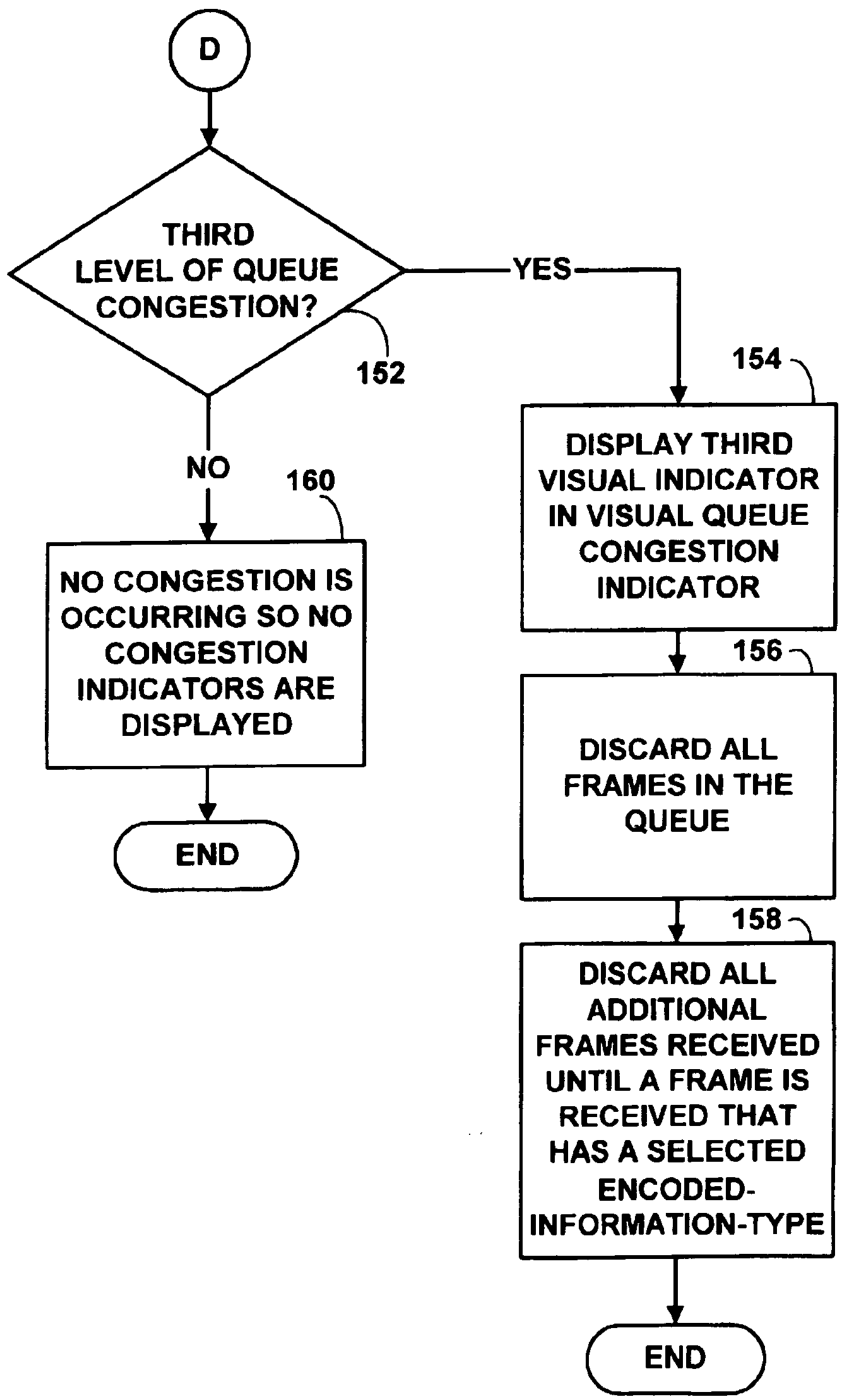

B
SECOND LEVEL OF QUEUE CONGESTION?
144
YES
NO
TO D FIG. 14C
146
DISPLAY SECOND VISUAL INDICATOR IN VISUAL QUEUE CONGESTION INDICATOR
148
SELECT AN ENCODED-INFORMATION-TYPE ENCODED IN A FRAME TO STORE IN THE QUEUE
150
DISCARD ALL ADDITIONAL FRAMES RECEIVED THAT DO NOT HAVE THE SELECTED ENCODED-INFORMATION-TYPE
END
C
140
SELECT AN ENCODED-INFORMATION-TYPE ENCODED IN A FRAME TO STORE IN THE QUEUE
142
DISCARD ALL FRAMES IN THE QUEUE THAT DO NOT HAVE SELECTED ENCODED-INFORMATION-TYPE
END

FIG. 14C


D
THIRD LEVEL OF QUEUE CONGESTION?
152
YES
NO
154
DISPLAY THIRD VISUAL INDICATOR IN VISUAL QUEUE CONGESTION INDICATOR
160
NO CONGESTION IS OCCURRING SO NO CONGESTION INDICATORS ARE DISPLAYED
END
156
DISCARD ALL FRAMES IN THE QUEUE
158
DISCARD ALL ADDITIONAL FRAMES RECEIVED UNTIL A FRAME IS RECEIVED THAT HAS A SELECTED ENCODED-INFORMATION-TYPE
END

METHOD AND APPARATUS FOR ADAPTIVE PRIORITIZATION OF MULTIPLE INFORMATION TYPES IN HIGHLY CONGESTED COMMUNICATION DEVICES

FIELD OF INVENTION

The present invention relates to communication in computer networks. More specifically, it relates to adaptive prioritization of multiple information types in highly congested communication devices.

BACKGROUND OF THE INVENTION

As is known in the art, a variety of computing devices are often connected together to form a computer network. The computer network may be a Local Area Network ("LAN") that connects devices over a small geographical area, or a Wide Area Network ("WAN") that connects devices over a large geographical area. The computing devices include video cameras, CD-ROMs, microphones, televisions, computers, modems, cable modems and other devices that send high resolution images, graphical images, moving images, audio and data in addition to textual information. Different types of computer networks may be interconnected to each other to form larger computer networks (e.g., the Internet). The interconnections include LAN-LAN, LAN-WAN, WAN-WAN, LAN-WAN-LAN, and other network interconnections.

The computing devices transfer multimedia information (e.g., audio, video and data) between two or more computer networks. Transferring multimedia information between two computer networks may or may not require a reserved bit rate transmission capacity, and a reserved bandwidth possibly for the total duration of the transaction. For example, multimedia information for a ten second video clip with sound that is being sent between two points in a Ethernet LAN, requires a significant portion of a 10 Mega-bits-per-second ("Mbps") data transmission capacity available on the LAN for ten seconds to send the multimedia information.

Gateways connect computer networks using different network protocols operating at different transmission capacities. For example, a gateway may have network connections to serial data lines connected to one or more modems. The serial data lines may be used one at a time at a relatively low transmission speed (e.g., 14,400 bps, 28,800 bps, or 56,000 bps) or can be bundled into a group at a higher transmission speed. In contrast, the gateway may also have one or more LAN connections (e.g., Ethernet, Token Ring, or Fiber Distributed Data Interface ("FDDI")). The LAN connections are higher speed connections (e.g., 10 Mbps) and are shared among multiple devices.

The gateway translates information contained in a first protocol being used on a first network connection into a second protocol being used on second network connection, and visa-versa, without undue delay or loss of information. For example, a modem operating at 28,800 bps may be using the International Telecommunications Union-Telecommunication Standardization Sector ("ITU-T", formerly known as the CCITT) H.324 audio/video conferencing protocol and a LAN operating at 10 Mbps may be using the ITU-T H.323 audio/video conferencing protocol for a video conferencing connection. A gateway translates H.323 from the LAN into H.324 for use on the modem, and visa-versa, without undue delay or loss of information even though the LAN is transmitting information at 10 Mbps and the modem is transmitting information at 28,800 bps.

However, the gateway may also translate H.323 on a LAN to H.323 on a serial line using the Point-to-Point Protocol

("PPP"), H.323 on a LAN to H.320 on an Integrated Services Digital Network ("ISDN") line, or translate H.32x on a LAN or a serial line to H.32x on a LAN or a serial line. The gateway is also responsible for maintaining timing relationships and packet sequencing even though a first protocol may use timing relationships and a second protocol may not use timing relationships.

When gateways are used to translate multimedia information between two computer networks, logical multimedia channels are typically created with separate audio, video and data channels. The audio and video channels are typically allocated with predetermined, fixed maximum bandwidth. For example, on a modem connection a audio channel may have a bandwidth of 5,300 bps and a video channel may have a bandwidth of 23,500 bps for a multimedia bandwidth of 28,800 bps. A LAN connection may use audio and video channels with larger bandwidth allocations since the LAN is capable of transmitting information at a much larger overall multimedia bandwidth (e.g., 10 Mbps).

There are several problems associated with using gateways or other internetworking devices known in the art to interconnect computer networks operating at different transmission capacities. For example, the logical channels for larger bandwidth computer network connections (e.g., LAN connections) are often connected to logical channels for smaller bandwidth computer network connections (e.g., modem connections). The larger bandwidth connections have no way of determining they are connected to smaller bandwidth, and more constrained connections. This will cause a constant congestion problem on a gateway since the logical channels for the larger bandwidth network is constantly transmitting more information than can be accepted by the lower bandwidth network. In addition, the gateway must translate between two or more different protocols for the connections without undue delay or loss of information.

Fig 2.
RTP(91
HANDLER
32
CODEC(82)

If data is sent along with the audio and video information on a multimedia connection, the congestion problems are further aggravated on the gateway. The gateway allocates a chunk of transmission bandwidth for a logical data channel. Since the data transmission is typically very bursty, the transmission bandwidth allocated for data channel is often wasted when no data is being sent. For example, the gateway may allocate a 5,000 bps logical data channel for each network connection using the data. For a modem with a bandwidth of 28,800 bps, this wastes about 17% of the available bandwidth on the modem. This is a significant waste of bandwidth on a smaller bandwidth network connection.

Another problem is that the gateway must maintain timing relationships and packet sequencing translating between certain protocols. The timing relationships and packet sequencing must be maintained even though a first protocol uses timing and a second protocol does not.

SUMMARY OF THE INVENTION

In accordance with a preferred embodiment of the present invention, the congestion problems for translating protocols are overcome. A method and apparatus for adaptively prioritizing between two or more encoded-information-types received over a communication link in multiple frames in an internetworking device (e.g., a gateway) is provided. The frames include multiple data bits. The method includes receiving multiple frames for a first network protocol over a first communication link having a first communication bandwidth. The multiple frames have one of multiple of encoded-information-types.

The received frames are stored in a delay variance removing queue in memory in the internetworking device. The delay variance removing queue allows the internetworking device to compensate for sequencing or timing relationships used in the first network protocol. The delay variance removing queue is also used as a buffer if it is not necessary to maintain timing relationships between two similar protocols. The frames from the delay variance removing queue are translated into a second network protocol. Periodically a test is completed to determine if a number of frames arriving on the first network connection exceeds a predetermined queue congestion threshold. The delay variance queue can be overflowed if the first communication link has a larger bandwidth than the second communication link. If the queue congestion threshold is exceeded, an encoded-information-type encoded in a frame is selected to store in the delay variance removing queue. Received frames are discarded that do have the selected encoded-information-type until enough frames have been processed so the delay variance removing queue has reached a predetermined length. The translated frames for the second network protocol are sent over a second communication link having a second communication bandwidth, the second communication bandwidth being less than the first communication bandwidth.

In another embodiment of the present invention, received frames that do not have the selected encoded-information-type are discarded and not stored in the delay variance removing queue. In yet another embodiment of the present invention, a new encoded-information-type may be selected for a specified time period (e.g., to send data) and all frames that do not have the new encoded-information-type are discarded during the specified time period.

The method and apparatus allow received frames with a selected-information-type to be discarded in a multimedia gateway without significantly affecting the quality of information when a first network protocol (e.g., H.323/LAN) is translated into a second network protocol (e.g., H.320, H.323/PPP, H.324, H.32x, etc.). The discarding of information is often necessary since frames are arriving on a communication link with larger bandwidth faster than they can be translated and sent on a communication link with smaller bandwidth.

The foregoing and other features and advantages of a preferred embodiment of the present invention will be more readily apparent from the following detailed description, which proceeds with references to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a computer network used to implement a preferred embodiment of the present invention;

FIGS. 2A and 2B are a flow diagram illustrating a method for adaptively prioritizing between two or more information types received on a communication link;

FIG. 3 is block diagram illustrating a packet synchronization in an internetworking device;

FIG. 4 is a block diagram illustrating a delay variance removing queue in an internetworking device;

FIG. 5 is a flow diagram illustrating a first method of discarding frames from a delay variance removing queue;

FIG. 6 is a block diagram illustrating the method of FIG. 5;

FIG. 7 is a flow diagram illustrating a second method of discarding frames from a delay variance removing queue;

FIG. 8 is a block diagram illustrating the method of FIG. 7;

FIG. 9 is a flow diagram illustrating a method for discarding received frames;

FIG. 10 is a block diagram illustrating the method of FIG. 9;

FIG. 11 is a flow diagram illustrating method for receiving frames with a new selected information type;

FIG. 12A is a block diagram illustrating a protocol translation system;

FIG. 12B is a block diagram illustrating another protocol translation system;

FIG. 13 is a block diagram illustrating a visual queue congestion indicator; and

FIGS. 14A, 14B and 14C are a flow diagram illustrating a method for displaying a visual indication of congestion in a queue.

DETAILED DESCRIPTION OF A PREFERRED EMBODIMENT

Protocol Translation System

FIG. 1 is a block diagram of a computer network **10** used to implement a preferred embodiment of the present invention. Computer network **10** includes a first computer network **12** and a second computer network **14** interconnected by an InterNetworking Device **16** ("IND"). However, more or fewer computer networks could be interconnected by IND **16** and the invention is not limited to interconnecting two computer networks. In addition, computer network **10** may include additional network devices (i.e., other than IND **16**) and additional network nodes which are not shown in FIG. 1.

IND **16** is also called an "InterWorking Unit" ("IWU"), an "Intermediate System" ("IS") or a "gateway." IND **16** has multiple communication links **18** to first computer network **12** and multiple communication links **20** to second computer network **14** (illustrated as single connections **18** and **20** in FIG. 1). There may also be multiple virtual communication channels over the communication links (**18**, **20**) (e.g., separate virtual channels for audio, video and data information). IND **16** has a software server **22** with a listing of network protocols **24** for first computer network **12** and a listing of network protocols **26** for second computer network **14**. IND **16** uses software server **22** to translate network protocols that arrive on a communication link from one computer network into a protocol for another computer network without undue delay or loss of information.

For example, a first network protocol **P1** that arrives over a first communication link **18** with a first communication bandwidth (e.g., 10 Mbps) from first computer network **12** is translated with software server **22** into a second network protocol **P2** for second computer network **14** using protocol listings **24** and **26**. Protocol **P2** is sent over a second communication link **20** to second computer network **14**. Second communication link **20** may have a second communication bandwidth (e.g., 28,800 bps) that is smaller than the first communication bandwidth but protocol **P2** is sent without undue delay or loss of information.

An operating environment for IND **16** of the present invention includes a processing system with at least one high speed Central Processing Unit ("CPU"), in conjunction with a memory system. Although described with one CPU, alternatively multiple CPUs may be used.

The memory system includes main memory and secondary storage. The main memory is high-speed Random Access Memory ("RAM") and Read Only Memory ("ROM"). Main memory can include any additional or alternative high-speed memory device or memory circuitry.

Secondary storage takes the form of long term storage, such as ROM, optical or magnetic disks, organic memory or any other volatile or non-volatile mass storage system. Those skilled in the art will recognize that the memory system can comprise a variety and/or combination of alternative components.

In accordance with the practices of persons skilled in the art of computer programming, the present invention is described below with reference to acts and symbolic representations of operations that are performed by the processing system, unless indicated otherwise. Such acts and operations are referred to as being "computer-executed" or "CPU-executed."

It will be appreciated that the acts and symbolically represented operations include the manipulation of electrical signals by the CPU. The electrical system represent data bits which cause a resulting transformation or reduction of the electrical signal representation, and the maintenance of data bits at memory locations in the memory system to thereby reconfigure or otherwise alter the CPU's operation, as well as other processing of signals. The memory locations where data bits are maintained are physical locations that have particular electrical, magnetic, optical, or organic properties corresponding to the data bits.

The data bits may also be maintained on a computer readable medium including magnetic disks, optical disks, and any other volatile or non-volatile mass storage system readable by the computer. The computer readable medium includes cooperating or interconnected computer readable media, which exist exclusively on the processing system or be distributed among multiple interconnected processing systems that may be local or remote to the processing system.

Adaptive Prioritization of Multiple Information Types in a Protocol

FIGS. 2A and 2B are a flow diagram illustrating a method **28** for prioritizing between two or more encoded-information-types for a selected network protocol received in frames over a communication link on an internetworking device such as IND **16**. The frames include multiple data bits. At step **30** in FIG. 2A, multiple frames are received for a first network protocol over a first communication link having a first communication bandwidth. The multiple frames include multiple encoded-information-types (e.g., video codec frames, audio codec frames and data frames). For selected protocol translations (e.g., H.323→H.324) that require timing be maintained, received frames are stored in a Delay Variance Removing ("DVR") queue in memory in the internetworking device at step **32**. The DVR queue allows IND **16** to compensate for packet sequencing and timing relationships during protocol translation. The frames from the DVR queue are translated into a second network protocol at step **34**. In a preferred embodiment of the present invention, the second communication bandwidth is less than the first communication bandwidth but the translated frames are sent without undue delay or loss of information. However, the communication bandwidths may also be equivalent on both the first and second communication links.

Periodically, a test is completed at step **38** (FIG. 2B) to determine whether a number of frames arriving on the first network connection exceeds a predetermined queue congestion threshold. If the congestion threshold is exceeded, an encoded-information-type encoded in a frame is selected at step **40** to store in the DVR queue. Frames that do not have the selected encoded-information-type are discarded at step **42** until enough frames in the DVR queue are processed so the DVR queue reaches a predetermined length. When enough frames in the DVR queue are processed so the DVR queue reaches a predetermined length at step **38**, discarding of frames that do not have the selected encoded-information-type is discontinued at step **44**. Translated frames for the second network protocol are sent over a second communication link having a second communication bandwidth at step **36** (FIG. 2A).

The first and second communication links (**18**,**20**) in method **28** may also include two or more virtual channels. For example, the two or more virtual channels may include a first virtual channel for audio information and a second virtual channel for video information, and a third virtual channel for data information.

In another embodiment of the present invention, if no timing compensation is necessary, then the received frames are stored in a DVR queue used as a buffer. When the DVR queue is used as a buffer, no timing compensation is completed in the DVR queue. The buffer is used to allow a protocol from a larger bandwidth connection to be translated into a protocol used over a smaller bandwidth connection (e.g., H.323/LAN→H.323/PPP).

For an embodiment using the DVR queue as a buffer, returning to FIG. 2A at step **30**, multiple frames are received for a first network protocol over a first communication link having a first communication bandwidth. The frames are stored in the buffer at step **32**. The frames stored in the buffer are translated into a second network protocol at step **34**. Periodically, a test is conducted at step **38** (FIG. 2B) to determine whether a number of frames arriving on the first network connection exceeds a predetermined buffer congestion threshold. If the buffer congestion threshold is exceeded, an encoded-information-type encoded in a frame is selected at step **40** to store in the buffer. Frames that do not have the selected encoded-information-type are discarded at step **42** until enough frames in the buffer are processed so the buffer reaches a predetermined length. When enough frames in the buffer are processed so the buffer reaches a predetermined length at step **38**, discarding of frames that do not have the selected encoded-information-type is discontinued at step **44**. Translated frames for the second network protocol are sent over a second communication link having a second communication bandwidth at step **36** (FIG. 2A).

As is known in the art, the Open Systems Interconnection ("OSI") model is used to describe computer networks. The OSI model consists of seven layers including from lowest-to-highest, a physical, data link, network, transport, session, application and presentation layer. The physical layer transmits bits over a communication link. The data link layer transmits error free frames of data. The network layer provides internetworking, determines routing information and controls a communications subnet. A communications subnet is a collection of transmission media required for routing and data transmission. Most INDs (e.g., gateways) operate at all levels in the OSI model.

Standards organizations such as the International Telecommunications Union-Telecommunication Standardization Sector ("ITU-T", formerly known as the CCITT), the Institute of Electrical and Electronic Engineers ("IEEE"), International Organization for Standards ("ISO") and others establish recommendations for standard network protocols. Devices connected to computer networks such as INDs typically use standard protocols as well as proprietary protocols to communicate with other devices in a computer network. INDs identify both standard and proprietary network protocols.

In one embodiment of the present invention, method **28** is used to translate audio/video conferencing protocols in a

multimedia gateway (e.g., IND 16) that are received for audio/video conferencing. First communication link 18 is a LAN communication link with an exemplary bandwidth of 10 Mbps. Second communication link 20 is a serial communication link via a modem with an exemplary bandwidth of up to 56,000 bps. However, the invention is not limited to the exemplary bandwidths, and other bandwidths may also be used for the first and second communication links (18, 20).

Method 28 is used in IND 16 with a DVR queue or with a DVR queue as a buffer for translating between ITU-T H.323 and H.323 (e.g., H.323/LAN and H.323/PPP), H.323 and ITU-T H.324, H.323 and ITU-T H.320 or ITU-T H.32x protocols, where "x" represents a protocol in the ITU-T H.320 series. As is known in the art, ITU-T H.324, entitled "Terminal for Low Bit Rate Multimedia Communication" is the ITU-T recommendation for standard videoconferencing using Plain Old Telephone Service ("POTS") lines. H.324 uses ITU-T H.223 for its data link layer. H.223 is entitled "Multiplexing Protocol for Low Bitrate Communications." ITU-T H.323, entitled "Visual Telephone Systems and Terminal Equipment for Local Area Networks That Provide a Non-Guaranteed Quality of Service," and H.323 version-2, entitled "Packet Based Multi-media Communication Systems," are the main family of video conferencing recommendations for Internet Protocol ("IP") networks. H.323 uses H.225, entitled "Media Stream Packetization and Synchronization on Non-Guaranteed Quality of Service LANs" or H.225 version-2 entitled "Call Signaling Protocols and Media Stream Packetization for Packet Based Multi-media Communication Systems" for its data link layer. ITU-T H.320, entitled "Narrowband Visual Telephone Systems and Terminal Equipment," is a standard protocol for videoconferencing using ISDN or similar telephone circuits.

For example, IND 16 using method 28 will identify H.323 sent by first computer network 12 over communication link 18 (a LAN communication link), translate H.323 data into H.320, H.324 or H.32x and send it to second computer network 14 over communication link 20 (a serial communication link, or a LAN communication link). IND 16 could also identify H.323 from second computer network 14 sent over communication link 20, translate H.323 data into H.320, H.324 or H.32x data, and send it to first computer network 12 over communication link 18. First computer network 12 and second computer network 14 would then have a two-way audio/video conferencing communication path.

During an audio/video conferencing call, audio information is typically supplied by audio Input/Output ("I/O") equipment (e.g., a microphone/speaker) that uses an audio coder/decoder ("codec") to capture audio information. Audio codecs are known to those skilled in the art and include G.711, G.722, G.723.1.1.1, G.728 and G.729. However, other audio codecs could also be used.

Video information is captured by video I/O equipment (e.g., a video camera) that uses a video codec. Video codecs are known to those skilled in the art and include H.261 and H.263 video codecs. However, other video codecs could also be used. ITU-T H.261, entitled "Video Codec for Audiovisual Services at Px64 Kbit/sec," defines video coding based on P-number of 64,000 bps channels (e.g., ISDN 64K bps B-channels) where P is two or more. H.261 is a video coding method designed for H.320, H.323 and other H.32x protocols. ITU-T H.263, entitled "Video Coding for Low Bit Rate Communication" is a video coding method used for H.323, H.324, and other H.32x protocols. H.263 uses the techniques of H.261 plus significant enhancements. For example, H.263 uses half pixel precision for motion compensation while H.261 uses full pixel precision and a loop filter. H.263 supports five resolutions including Quarter Common Interchange Format ("QCIF"), Common Interchange Format ("CIF"), 4CIF, 16CIF, which are four and sixteen times the resolutions of CIF, and Sequenced Quarter Common Interchange Format ("SQCIF"). H.261 supports only QCIF and CIF.

H.263 uses three types of video codec frames as is shown in Table 1 below.

TABLE 1

H.263 Codec Frame Types	Description
I-Frame	I-frames are intra frames which are encoded similarly to Joint Picture Expert Group ("JPEG"). This means, no motion compensation prediction is used. These frames are sent periodically (e.g., every second) to enable a decoder to recover from errors. The frame size of an I-frame can be 5K bits to 50K bits for Quarter Common Interchange Format ("QCIF") and larger for Common Interchange Format ("CIF").
P-Frame	P-frames are error values for the motion compensation prediction process that is used. If the motion prediction process is good, P-frames consist of motion vectors with no error data. If the motion compensation prediction process is bad the error data is usually between 100 bits and 10K bits of data.
B-Frame	B-frames are predicted using a previous I-frame or P-frame and a current P-frame. The main use of B-frames is the ability to increase the frame rate with only a minimal amount of increase in the bit rate. B-frames are not sent alone but are combined with P-frames.
PB-Frame	PB-frames are P-frames jointly encoded with a B-frame. The PB-frames are two frames and there is a delay of one frame involved. PB-frames average about 200 bits to 20K bits of data.

P, and PB frames are used to lower the total amount of bits transmitted to display a video image by sending motion compensation prediction errors. I-frames are sent periodically to recover from the motion compensation prediction of the P, and PB frames. Sending I-frames is necessary but not desirable since the I-frame contains bits representing the whole video image while P and PB frames contain bits representing only changes in the video image since the last frame (I, P, or PB) was sent. For example, an I-frame may contain 5000–50,000 bits of information, while a P-frame may contain 100- 10,000 bits of information. The P, and PB frames can be transmitted faster and reduce the congestion on the communication link.

The H.245 and T.120 protocols and Point-to-Point Protocol ("PPP") are also used during audio/video conferencing. H.245, entitled "Control Protocol for Multimedia Communication," defines initiation of communications between systems and negotiation of capabilities procedures.

T.120, entitled "User Data Transmission Using a Multi-Layered Protocol ("MLP") is the ITU-T recommendation for data conferencing. PPP is used to encode network layer data over a serial communication link. For more information on H.320, H.323, H.324, H.261, H.263, H.245 and T.120 see "*Mainstream Videoconferencing: A Developer's Guide to Distance Multimedia*" by Joe Duran and Charlie Sauer, Addison, Wesley, Reading, Mass., 1997 or "*Videoconferencing and Videotelephony: Technology and Standards*", by Richard Schaphorst, Artech House, Norwood, Mass., 1996.

For more information on PPP see *Internet Request For Comments* ("RFC") 1060. The above documents are expressly incorporated herein by reference.

Audio, video, data and control information (G.7xx, H.261, H.263, H.245, T.120) is encoded in H.225 frames for use with H.323 and H.223 frames for use with H.324. The H.225 and H.223 frames are data link frames used in the OSI data link layer over first and second communication links (**18**,**20**). The video codec frames (J, P, and PB) are encoded in H.225 or H.223 frames. The multimedia information encoded in the H.225, H.223 or other frames is hereinafter referred to as "encoded-information-types."

On the larger bandwidth communication link **18** using H.323 (i.e., LAN or other network side) contention losses, retries and unreliable links indicate protocol information packets may be lost or delayed an indeterminable amount of time. Thus, first computer network **12** may use Real-Time Protocol ("RTP") in its protocol stack to provide a timing synchronization and sequence number mechanism when H.323 is used. RTP allows the H.323 to compensate for packets varied in time and lost packets. However, H.323 may use other protocols for packet timing and sequencing.

The H.324 protocol typically does not provide any timing relationships like those provided with RTP in H.323. Thus, IND **16** provides packet synchronization after translation before sending the packet to the H.324 side. A DVR queue in IND **16** is used to provide packet synchronization and sequencing. Packet and time sequencing is not required for H.323-to-H.323 (e.g., H.323/LAN→H.323/PPP) translation, and DVR queue is used as a buffer in IND **16** for such an embodiment.

FIG. **3** is a block diagram **46** illustrating packet synchronization in IND **16**. As is shown in FIG. **3**, IND **16** receives H.323 packet information (e.g., audio, video and data information) from first computer network **12**, over first communication link **18**. First communication link **18** is a larger bandwidth communication link (e.g., a 10 Mbps LAN communication link) than second communication link **20** (e.g., a 56,000 bps serial line communication link). The H.323 packets are sent as H.225 frames over communication link **18** to IND **16**. IND **16** translates H.323 into H.324 and sends H.324 information to second computer network **14** over second communication link **20** as H.223 frames. H.324 and H.320 provide timing and sequencing relationships at a source. However, H.324 or H.320 do not provide any in-stream timing as is provided by RTP in H.323. H.324 and H.320 align data streams at a source and then "assume" that the underlying transmission medium will not cause an "jitter" (i.e., timing variations in the transmitted frames). As a result, IND **16** provides packet timing synchronization and packet sequencing in the DVR queue before sending H.324 data packets to second computer network **14**. IND **16** removes transmission "jitter" as well as provides "lip synchronization" for H.323→←H.324 translation. For, H.323/LAN→←H.323/PPP translation, no timing synchronization is required, so IND **16** does not provide either jitter removal or lip synchronization, but does provide the data rate adaptation described above (i.e., 10 Mpbs→←56,000 bps or other modem speeds).

As is shown in FIG. **3**, first computer network **12** sends six H.323 protocol packets numbered **P1**, **P2**, **P3**, **P4**, **P6**, and **P5** to IND **16** over first communication link **18**. The packets would be received in H.225 frames. Packet **P6** was sent before packet **P5** (i.e. is out of sequence). The packets are sent in a "bursty" manner as is illustrated by the timing marks T1–T6. IND **16** accepts the packets in H.225 frames and stores them in the DVR queue.

IND **16** translates the packets from the DVR queue into H.324 protocol, and synchronizes the packets for output on second communication link **20** to second computer network **12** (e.g., provides audio/video "lip" synchronization). The H.324 packets are output as H.223 frames using the sequence numbers and the appropriate timing as is illustrated by timing marks T1'–T6'. Packets **P1**, **P2** and **P3** are output on timing marks T1' T2' and T3' respectively. Packets **P6** and **P5** have been re-ordered in the proper sequence and output on timing marks T5' and T6' respectively. Thus, H.324 packets are output with the proper timing and packet sequencing by IND **16**. IND **16** gives H.320 and H.324 a constant stream of output packets even though the H.320 and H.324 do not include timing values with each packet. No audio/video synchronization or "lipsync" is required for H.323-to-H.323 translation (e.g., H.323/LAN→H.323/PPP).

FIG. **4** is a block diagram **48** illustrating a DVR queue **50** used in IND **16**. DVR queue **50** illustrates the packet sequence received from first computer network **12** in FIG. **3**. In a preferred embodiment of the present invention, DVR queue **50** is a predetermined fixed length. If first communication link **18** is a larger bandwidth than second communication link **20**, DVR queue **50** may overflow and frames containing the packets are discarded as is illustrated in method **24**. Software server **22** in IND **16** uses DVR queue **50** to provide packet sequencing and packet timing synchronization when translating between two network protocols (e.g., H.323 and H.324) that require timing synchronization.

Method **28** (FIGS. **2A** and **2B**) is illustrated with one embodiment of the present invention. First computer network **12** (FIG. **1**) uses H.323 and sends H.225 frames with multimedia information (e.g., audio, video and data information) as encoded-information-types to IND **16** over first communication link **18** which has first bandwidth (e.g., 10 Mbps). Second computer network **14** uses H.324 and receives H.223 frames with the encoded-information types from IND **16** over second communication link **20** which has a second smaller bandwidth (e.g., 28,800 bps) without undue delay or loss of information. However, the present invention is not limited to this illustrative embodiment and other embodiments may be used and the H.323 protocol can be translated into H.320 and H.323 by IND **16**.

At step **30** (FIG. **2A**) of method **28**, multiple H.225 frames are received for the H.323 protocol over first communication link **18**. The multiple frames include encoded multimedia information (e.g., encoded audio, video and data information) as encoded-information-types. The H.225 frames include multiple video codec encoded-information-types shown in Table 1. The received H.225 frames are stored in DVR queue **50** (FIG. **4**) in a memory of IND **16** at step **32**. The H.225 frames with H.323 data from DVR queue **50** are translated into H.223 frames for H.324 at step **34**. DVR queue **50** is used to provide packet sequencing and packet timing synchronization. The translated H.223 frames are sent over second communication link **20** at step **36**.

Periodically (e.g., once per second), a test is completed at step **38** (FIG. **2B**) to determine whether the number of H.225 frames containing multimedia information arriving on first network connection **18** exceeds a predetermined queue congestion threshold for DVR queue **50**. The queue congestion threshold is determined based on the size of DVR queue **50**. If the queue congestion threshold is exceeded, an encoded-information-type is selected at step **40** (e.g., H.263 I-frames encoded in H.225 frames).

For example, H.263 I-frames are video codec frames without any motion compensation predictions, and H.263

P-frames, frames, and PB-frames include errors for motion compensation predication. The H.263 P, and PB frames can be discarded and only the H.263 I-frames encoded in H.225 frames are used to transmit audio/video conferencing data. I-frames are sent periodically and allow a video decoder to recover from any motion prediction errors generated by P and PB frames.

H.225 frames that do not have the selected encoded-information-type (e.g., H.225 frames with encoded P, and PB frames) are discarded at step **42** until the number of frames arriving on first communication link **18** is less than the predetermined queue congestion threshold. H.225 frames with the selected encoded-information-type (e.g., H.263 I-frames) are stored in DVR queue **50** at step **32** to allow translation between H.323 and H.324 at step **34**. When enough frames in the DVR queue are processed so the DVR queue reaches a predetermined length at step **38**, discarding of frames that do not have the selected encoded-information-type is discontinued at step **44**.

Since one of the communication links typically has a larger communication bandwidth than the other communication link, method **24** is used to compensate for the differences in communication bandwidth, timing and sequencing without undue delay or loss of information. Method **28** is described with respect to video codec information (i.e., H.263 video code I, P, and PB frames). However, method **28** can also be used with audio information from the audio codecs described above (e.g., selecting between audio codec data for lower and higher fidelity audio) and with data information. Method **28** can also be used for translating protocols that do not require timing synchronization with a DVR queue.

In an another embodiment of the present invention, X-number of frames which do not have the selected encoded-information-type are discarded at step **42** until Y-number of frames with the selected encoded-information-type are stored in DVR queue **50** at step **32** before checking the queue congestion threshold again at step **38**.

In another embodiment of the present invention, a flow control message is sent by IND **16** over first communication link **18** to first computer network **12** to practice frame discarding at step **42**. In such an embodiment, a flow control message is sent via the H.245 protocol and requests first computer network **12** send frames with encoded-information-types at a rate less than the available bandwidth on first communication link **18** so more frames are processed. A timer is set on IND **16** with a predetermined congestion control (e.g., 1 second) value to allow IND **16** to process the frames stored in DVR queue **50**. When the timer expires, a second control flow message is sent with H.245 protocol to inform first computer network **12** that it may send frames with encoded-information-types again using the available bandwidth on first communication link **18**.

Adaptive Prioritization of Multimedia Protocols

In a preferred embodiment of the present invention, selected frames are discarded at step **42** from DVR queue **50** with method **52**. FIG. **5** is a flow diagram illustrating a method **52** for discarding received frames from DVR queue **50**. At step **54**, a first location in DVR queue **50** is determined containing a first frame with the selected encoded-information-type (e.g., H.225 frame with an encoded H.263 I-frame). At step **56**, a second location is determined containing a second frame with the selected encoded-information-type. The frames (P, and multiple PB frames) between the first and second locations are discarded at step **58**, thereby increasing space in DVR queue **50**. As was discussed above, encoded H.263 P, and PB frames can be discarded since encoded I-frames alone can be used to display video information. If a first location in DVR queue **50** containing a first frame with the selected encoded-information type cannot be found at step **54**, frames are discarded from the DVR queue until a frame with the selected encoded-information type arrives. Method **52** can also be used to discard frames from a buffer used for translating protocols that do not require timing synchronization with a DVR queue.

FIG. **6** is a block diagram **60** illustrating method **52**. First location **62** (FIG. **6**) determined at step **54** (FIG. **5**) contains a first I-frame encoded in a H.225 frame. Second location **64** (FIG. **6**) determined at step **56** (FIG. **5**) contains a second encoded I-frame. The encoded P, and multiple PB frames between first and second locations (**62**,**64**) are discarded at step **58** thereby increasing space in DVR queue **50**.

In another embodiment of the present invention, selected frames are discarded at step **42** from DVR queue **50** with method **66**. FIG. **7** is a flow diagram illustrating a method **66** for discarding received frames from the DVR queue **50**. At step **68**, a first location containing a first frame with the selected encoded-information-type (e.g., H.225 frame with an encoded I-frame) is determined. At step **70**, frames in locations before first location **62** in the queue are discarded thereby increasing space in DVR queue **50**. Method **66** can also be used to discard frames from a buffer used for translating protocols that do not require timing synchronization with a DVR queue.

FIG. **8** is a block diagram **72** illustrating method **66**. At step **68** (FIG. **7**), a first location **74** (FIG. **8**) is determined containing a H.225 frame with an encoded I-frame. At step **70** (FIG. **7**), the H.225 frames with encoded P, and multiple PB frames before first location **74** (FIG. **8**) are discarded thereby increasing space in DVR queue **50**.

In another embodiment of the present invention, selected frames are discarded at step **42** as they are received with method **76** and are not stored in DVR queue **50**. FIG. **9** is a flow diagram illustrating method **76** for discarding frames as they are received. At step **78**, an average time period is determined when a frame with a selected encoded-information-type is received (e.g., a H.225 frame with an encoded H.263 I-frame). At step **80**, a timer is set with the average time period. At step **82**, frames are discarded until the timer expires or a frame with the selected encoded-information-type is received. Method **76** can also be used to discard frames from a buffer used for translating protocols that do not require timing synchronization with a DVR queue.

FIG. **10** is a block diagram **84** illustrating the method of FIG. **9**. It is determined at step **78** that a H.225 frame with a selected encoded-information-type (e.g., an I-frame) is received every 0.5 seconds at locations **86**, **88** and **90** corresponding to time periods of zero, 0.5 and one second. A timer would be set with the average time period of 0.5 seconds step **80**. Frames are discarded until the timer expires or a frame with the selected encoded-information-type is received at step **82**. The average time period determined is typically determined for a longer time period than is shown in FIG. **10** (e.g., over 1–2 minutes). The average time period may also be adjusted dynamically as the content of the audio/video conferencing information changes.

In another embodiment of the present invention, a request may be made by a computer network to select a new encoded-information-type for a specified time period (e.g., to send data) and discard all other frames with other encoded-information-types received during the specified time period.

FIG. 11 is a flow diagram illustrating method 92 for receiving frames with a new encoded-information-type. At step 94, a selection input is received from a computer network indicating a time period for continuously receiving frames with a new selected encoded-information-type. For example, IND 16 may receive a selection input from first computer network 12 for establishing a virtual data channel for T.120 data. T.120 could be used to send a spreadsheet or other collection of information data during the audio/video conference. At step 96, a timer is set with the indicated time period received in the selection input. At step 98, other frames received with encoded information (e.g., H.263 I, P, and PB frames) are discarded that do not have the new selected encoded-information-type (e.g., T.120) until the timer expires.

Adaptive Prioritization Systems

FIG. 12A is a block diagram illustrating a protocol translation system 100 for one illustrative embodiment of the present invention. A first multimedia computer 102 is connected to first computer network 12, which is a LAN. First computer network 12 is connected to IND 16 with first communication link 18 (e.g., a LAN connection at 10 Mbps). Multimedia computer 102 is using H.323 protocol for audio/video conferencing with a second multimedia computer 104 using H.324 protocol. Second multimedia computer 104 is connected to a second computer network 14, which is a serial communications device (e.g., a modem). Second computer network 14 uses a second communication link 20 (e.g., a serial telephone line at 56,000 bps, an ISDN line, etc.) to connect to IND 16.

IND 16 includes a LAN interface 106, a audio/video synchronization device 108 and a modem interface 110. Multimedia information (voice, video and data) is sent from multimedia computer 102 from first computer network 12 to IND 16 over communication link 18 as H.225 frames. Communication link 18 may have multiple virtual channels for audio, voice and data information. LAN interface 106 in IND 16 sends 112 audio information via RTP and sends 114 video information via RTP to audio/video synchronization device 108. RTP was explained above. T.120 data is sent 116 from LAN interface 106 to software server 22 using Transmission Control Protocol ("TCP") or another data transmission protocol known in the art. However, TCP may or may not terminate on IND 16 depending on the type of data protocol being used. In addition, other protocols could also be used to send audio, video or data information.

Audio/video synchronization device 108 is used synchronize audio and video information so audio information corresponds to a video image. For example, loss of lip synchronization for an audio/video conferencing call between two people is highly undesirable, so audio and video information is synchronized. In one embodiment of the present invention, audio/video synchronization device 108 uses DVR queue 50 (FIG. 4) along with software comprising packet sequencing and packet timing synchronization functionality. DVR queue 50 is used to provide packet sequencing and packet timing synchronization when translating between two network protocols as was explained above and illustrated in FIGS. 3 and 4.

Software server 22 uses method 28 and one or more of methods 52, 66, 76 and 92 described above to translate H.323 protocol information into H.324 protocol information and uses protocol translation tables 24 and 26 (FIG. 1). Software server 22 receives synchronized audio/video data over connection 118. Software server 22 uses audio codec Protocol Data Units ("PDUs") such as PDUs for the G.723.1 audio codec known in the art to send 120 synchronized audio

information to modem interface 110. However, other audio codec PDUs could also be used (e.g., G.711, G.722, G.728, or G.729). Software server 22 uses a video codec PDU such as a H.263 video codec PDU to send 122 synchronized video information to modem interface 110. However, other video codec PDUs could also be used (e.g., H.261). Software server 22 sends 124 any T.120 data received on TCP input 118 to modem interface 110 as data PDUs for a LAPM or other modem interface protocol known in the art. However, other protocols could also be used (e.g., V.14).

Software server 22 translates H.323 protocol information encoded in H.225 frames into H.223 frames with H.324 protocol information using audio/video synchronization device 108 and data input 116 and sends the H.324 protocol information as H.223 frames to modem interface 110. Modem interface 110 outputs H.324 protocol to first computer network 14 (i.e., modem) which is connected to multimedia computer 104. Multimedia computer 104 can also initiate an audio/video conference with multimedia computer 102, and the audio/video conferencing communication just described would be executed in reverse order (i.e., H.324→H.323).

FIG. 12B is a block diagram illustrating a protocol translation system 125 for a preferred embodiment of the present invention. A first multimedia computer 102 is connected to first computer network 12 that is a LAN. First computer network 12 is connected to IND 16 with first communication link 18 (e.g., a LAN connection at 10 Mbps). Multimedia computer 102 is using H.323/LAN protocol for audio/video conferencing with a second multimedia computer 104 using H.323/PPP protocol. Second multimedia computer 104 is connected to a second computer network 14, which is a serial communications device (e.g., a modem). Second computer network 14 uses a second communication link 20 (e.g., a serial telephone line at 56,000 bps, an ISDN line, etc.) to connect to IND 16. No audio/video synchronization is required to translate between H.323/LAN and H.323/PPP so audio/video synchronization device 108 and DVR queue 50 are replaced by buffer 107. LAN interface 106 in IND 16 sends 112 audio information via RTP and sends video information 114 via RTP to buffer 107. T.120 data is optionally sent 116 from LAN interface 106 to buffer 107 or directly to software server 22 with TCP or another transmission protocol. Audio/video stream and optional data stream 119 is sent from buffer 107 to software server 22 for translation. Software server 22 translates H.323/LAN into H.323/PPP protocol information using and sends the H.323/PPP protocol information to modem interface 110. Modem interface 110 outputs the H.323/PPP protocol to first computer network 14 (i.e., modem) which is connected to multimedia computer 104. Multimedia computer 104 can also initiate an audio/video conference with multimedia computer 102, and the audio/video conferencing communication just described would be executed in reverse order (i.e., H.323/PPP→H.323/LAN).

Visual Congestion Indicators

In one embodiment of the present invention, IND 16 includes a visual indication of any congestion in DVR queue 50 or a buffer used in place if DVR queue 50. FIG. 13 is a block diagram illustrating a visual queue congestion indicator 126. Visual Queue congestion indicator 126 includes a first congestion indicator 128, a second congestion indicator 130 and a third congestion indicator 132 which are three different colors (e.g., green, yellow and red). Only one congestion indication 128, 130 or 132 is displayed at any instance of time. For example, in FIG. 13, only second congestion indicator 130 is displayed. The congestion indi-

cators are used as a visual indication of the amount of congestion in DVR queue **50** or buffer and can be implemented in hardware as Light Emitting Diodes or other hardware components on IND **16** or in software as graphical shapes on a display associated with IND **16**. The visual congestion indicators (**128,130,132**) are not limited to the circle shapes shown in FIG. **13**.

For example, the first congestion indicator **128** (e.g., a green color) may indicate no congestion or a small amount of congestion. In one embodiment of the present invention, when first congestion indicator is displayed, methods **52** and **66** are used by IND **16** to discard frames from DVR queue **50** or buffer thereby increasing space in the DVR queue or buffer. When second congestion indicator **130** is displayed (e.g., a yellow color) an intermediate amount of congestion is occurring in DVR queue **50** or buffer. IND **16** uses method **78** to discard frames until a timer expires or a frame with a selected encoded-information-type is received. For example, all H.263 P, and PB frames encoded in H.225 are discarded until an H.263 I-frame is received. When third congestion indicator **132** is displayed (e.g., a red color), a major amount of congestion is occurring in DVR queue **50** or buffer. IND **16** sends a H.245 control message to ask a computer network to send frames at less than the available bandwidth for a determined period of time. IND **16** may also drop all frames received until the frames in DVR queue **50** or buffer are processed.

FIGS. **14A**, **14B** and **14C** are a flow diagram illustrating a method **134** for displaying a visual indication of congestion in DVR queue **50** or buffer. At step **136** (FIG. **14A**) a test is conducted to determine if a first level of congestion is occurring in DVR queue **50** or buffer. If so, at step **138**, first visual indicator **128** is displayed (e.g., with a green color). An encoded-information-type encoded in a data frame is selected at step **140** of FIG. **14A** (e.g., H.263 I-frame encoded in a H.225 frame). All frames in DVR queue **50** or buffer that do not have the selected encoded-information-type are discarded at step **142**.

If the first level of congestion is not occurring at step **136**, at second test is conducted at step **144** (FIG. **14B**) to determine if a second level of congestion is occurring. If so, at step **146** second visual indicator is displayed (e.g., with a yellow color). An encoded-information-type encoded in a frame is selected at step **148**. All additional frames received are discarded that do not have the selected encoded-information-type at step **150** (e.g., all H.263 P, and PB frames). Frames that do have the selected encoded-information-type are added to DVR queue **50** or buffer. The frames already in DVR queue **50** or buffer are processed as new received frames are being discarded.

If the second level of congestion is not occurring at step **144**, a third test is conducted at step **152** (FIG. **14C**) to determine if a third level of congestion is occurring. If so, at step **154** third visual indicator **132** is displayed (e.g., with a red color). An encoded-information-type encoded in a frame is selected at step **156**. All frames in DVR are discarded in DVR queue **50** or buffer at step **158**. All additional frames received are discarded at step **158** until a frame with the selected encoded-information-type is received.

If the third level of congestion is not occurring at step **152**, then no congestion indicators are displayed at step **160** indicating no congestion in DVR queue **50** or buffer. Method **134** and visual queue congestion indicator **136** are used in IND **16** to allow a user to visually determine the amount of congestion in DVR queue **50** or a buffer. Method **134** and visual queue congestion indicator **136** are beneficial to indicate congestion in DVR queue **50** or buffer when a first communication link connected to IND **16** has very large bandwidth and a second communication link has a smaller bandwidth (e.g., 10 Mbps LAN communication link and a 28,800 bps serial communication link). Method **134** and visual queue congestion indicator **136** can also be used with other queues or buffers that are not implemented as DVR queue **50**.

In one specific embodiment of the present invention, IND **16** (FIG. **12**) is implemented in a Total Control Enterprise Network Hub commercially available from 3Com Corporation of Santa Clara, Calif. The Total Control product includes multiple network interface cards connected by a common bus. See "Modem Input/Output Processing Signaling Techniques", U.S. Pat. No. 5,528,595, granted to Dale M. Walsh et al. for a description of the architecture of the Total Control product, which is incorporated by reference herein. For example, LAN interface **106** is a 3Com Ethernet or Token Ring interface card other LAN interface card. Modem interface **110** is a 3Com Quad Modem card (or the equivalent). Computer software server **22** is added to existing software in the Total Control product (such as in the Edgeserver card) to implement method **28** and one or more of methods **52**, **66**, **76**, **92** and **134** described above to accomplish features described herein. However, IND **16** can also be implemented in other devices with other hardware and software configurations and is of course not limited to implementation in a Total Control product or the equivalent.

It should be understood that the programs, processes, methods and apparatus described herein are not related or limited to any particular type of computer apparatus (hardware or software), unless indicated otherwise. Various types of general purpose or specialized computer apparatus may be used with or perform operations in accordance with the teachings described herein.

In view of the wide variety of embodiments to which the principles of the invention can be applied, it should be understood that the illustrated embodiments are exemplary only, and should not be taken as limiting the scope of the present invention. For example, the steps of the flow diagrams may be taken in sequences other than those described, and more or fewer elements may be used in the block diagrams.

The claims should not be read as limited to the described order or elements unless stated to that effect. In addition, use of the term "means" in any claim is intended to invoke 35 U.S.C. §112, paragraph 6, and any claim without the word "means" is not so intended. Therefore, all embodiments that come within the scope and spirit of the following claims and equivalents thereto are claimed as the invention.

We claim:

1. A method of adaptively prioritizing between two or more encoded-information-types received over a communication link in a plurality of frames in an internetworking device, the frames including a plurality of data bits, the method comprising the following steps:

receiving a plurality of frames for a first network protocol over a first communication link having a first communication bandwidth, the plurality of frames having one of a plurality of encoded-information-types;

storing the received frames in a delay variance removing queue in a memory in the internetworking device, the delay variance removing queue allowing the internetworking device to compensate for sequencing or timing relationships used in the first network protocol;

translating the frames from the delay variance removing queue into a second network protocol;

determining periodically whether a number of frames arriving on the first network connection exceeds a predetermined queue congestion threshold, and if so,

selecting an encoded-information-type encoded in a frame to store in the delay variance removing queue;

discarding received frames that do not have the selected encoded-information-type until enough frames in the delay variance removing queue have been processed so the delay variance removing queue reaches a predetermined length; and

sending the translated frames for the second network protocol over a second communication link having a second communication bandwidth, the second communication bandwidth being less than the first communication bandwidth.

2. A computer readable medium having stored therein instructions for causing a central processing unit to execute the method of claim 1.

3. The method of claim 1 wherein the step of discarding received frames includes:

determining a first location in the delay variance removing queue containing a first frame with the selected encoded-information-type;

determining a second location in the delay variance removing queue containing a second frame with the selected encoded-information-type; and

discarding other frames in other locations in the delay variance removing queue between the first and second locations, thereby increasing space in the delay variance removing queue.

4. The method of claim 1 wherein the step of discarding received frames includes:

determining a first location in the delay variance removing queue containing a first frame with the selected encoded-information-type; and

discarding other frames in locations in the delay variance removing queue prior to the first location, thereby increasing space in the delay variance removing queue.

5. The method of claim 1 wherein the step of discarding received frames includes:

discarding frames that do not have the selected encoded-information-type until X-number of frames have been received or a frame with the selected encoded-information-type is received.

6. The method of claim 1 wherein the step of discarding received frames includes:

sending a first flow control message over the first communication link requesting frames be sent at less than the first communication bandwidth;

setting a timer with a predetermined congestion control value; and

upon expiration of the timer, sending a second flow control message over the first communication link requesting frames again be sent at the first communication bandwidth.

7. The method of claim 6 wherein the first and second flow control messages are sent with the H.245 control protocol for multimedia communication.

8. The method of claim 1 wherein the step of discarding received frames includes:

determining a time period when a frame with the selected encoded-information-type is received;

setting a timer with the time period; and

discarding frames received until the timer expires or a frame with the selected encoded-information-type is received.

9. The method of claim 1 further comprising:

receiving a selection input with a time period for continuously receiving frames with a new encoded-information-type;

setting a timer with the time period; and

discarding frames received that do not have the new encoded-information-type until the time expires.

10. The method of claim 1 wherein the first communication link is communication link to a local area network communication link and the second communication link is a serial line communication link.

11. The method of clam 1 wherein the first network protocol is H.323 and the second network protocol is any of H.320, H.323 over a LAN, H.323 over PPP or H.324.

12. The method of claim 1 wherein the step of selecting an encoded-information-type includes selecting a H.263 I-frame as the encoded-information-type.

13. The method of claim 1 wherein the discarding step includes discarding received frames having an encoded-information-type of any of H.263 P-frames, frames or PB-frames.

14. The method of claim 1 wherein the step of discarding received frames includes:

determining whether a first frame with the selected encoded-information-type is in the delay variance removing queue, and if not,

discarding received frames that do not have the selected encoded-information-type until a frame with the selected encoded-information-type is received.

15. The method of claim 1 wherein the first and second communication links have two or more virtual channels for sending and receiving frames and the two or more virtual channels include a first virtual channel for audio information, a second virtual channel for video information and a third channel for data information.

16. The method of claim 1 wherein the plurality of frames received over the communication link include any of H.225 or H.223 frames.

17. The method of claim 1 wherein the delay variance removing queue is a buffer that is used to translate network protocols that do not require timing synchronization with a delay variance removing queue.

18. An internetworking apparatus for translating a plurality of frames received for a first network protocol over a first communication link having a first communication bandwidth, the plurality of frames having one of a plurality of encoded-information-types into a second network protocol sent over a second communication link having a second communication bandwidth, the apparatus comprising:

a delay variance removing queue, for allowing the internetworking device to compensate for sequencing or timing relationships used in a first network protocol;

a translator, for translating frames from the first network protocol stored in the delay variance removing queue into a second network protocol; and

a queue congestion threshold for discarding received frames that do have a selected encoded-information-type until enough frames in the delay variance removing queue are processed so the delay variance removing queue reaches a predetermined length.

19. The internetworking apparatus of claim 18 wherein the delay variance removing queue is a buffer that is used to translate network protocols that do not require timing synchronization with a delay variance removing queue.

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